The acoustics of small rooms at low frequencies.

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THE ACOUSTICS OF SMALL ROOMS AT LOW FREQUENCIES

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by

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Abstract

This thesis is an examination of the causes and prevention of colourations of speech in a small room, which are particularly noticeable when transmitted by a microphone and loudspeaker. Previous work by the author is summarised in a paper (Gilford, 1959) which is bound in at the end. After a general statement of the problems still remaining for solution an attempt is made to find whether image array analysis can be applied to non-rectangular as well as to rectangular rooms. As the problem is intractable, experiments were made on models. They showed that a room deviating from rectangular shape by angles of up to about five degrees has modes up to a high order identical in frequency with those of a rectangular room of the same mean dimensions.

Means of assessing colourations and of displaying the objective features which accompany them are examined and a method using a narrow-band speech spectrograph is proposed. The design of the filter sections is described but the equipment has not yet reached completion.

The properties of several types of resonant low-frequency sound absorbers are next compared as a means of reducing colourations. It is concluded that membrane absorbers are the most effective for the provision of general low-frequency absorption in designing acoustic treatment, but Helmholtz resonators of narrow bandwidth are the most suitable for specific remedial

action. A theory of functional absorbers is presented.

Theoretical and experimental work show little potential advantage in using absorbers in the edges or corners of a room where the reverberant sound pressure is highest.

The importance of diffusion is stressed and correct distribution of absorbers is shown to be a better method of improving diffusion then irregularities in room shape.

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THE ACOUSTICS OF SMALL ROOMS AT LOW FREQUENCIES.

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THE ACOUSTICS OF SMALL ROOMS AT LOW FREQUENCIES

CHAPTER 1

INTRODUCTION

The most serious and intractable of the defects of small broadcasting studios, such as those used for talks, news or the performance of music by small combinations of instruments, are the so-called colourations at low frequencies.

A colouration may be defined as a subjective effect appearing as a selective reinforcement of sound of a particular frequency in relation to the spectrum as a whole, occurring in the process of transmission of the sound through the studio from the performer to the microphone or to a listener in the studio.

This effect is absent in a room which is substantially free from reflections by reason of the treatment of its surfaces with sound-absorbing materials, and it must therefore presumably be associated with the reflected sound which constitutes the reverberant sound-field.

This association is commonly accepted and is borne out by all experimental evidence, but no direct correlation has yet been established between physically measurable properties of the sound-field and the subjective impression of colouration.

Certainly, colourations must arise from the frequency-variations of some parameter of the total sound-field, but an examination of the variations of all measurable properties such as pressure or particle velocity over the audible spectrum does not give immediate support to this hypothesis. When a room is excited by a constant strength source, these variables all fluctuate rapidly with frequency, between maxima and minima which may be substantially unchanged over a wide frequency range, while it is not normally possible to hear and recognise more than two or three colouration frequencies from the same room. There is, therefore, no simple one-to-one correspondence between colouration frequencies and, say, pressure maxima, and it is therefore necessary to seek the explanation elsewhere.

Very little systematic work has been carried out on this subject, since it is of interest almost exclusively to broadcasting and sound recording organisations. The study of acoustics as it affects the naturalness or aesthetic quality of the sound heard by those actually present in the same enclosure as the source of sound has been principally concerned with large auditoria such as concert halls and cinematograph theatres; in so far as small rooms such as lecture rooms or conference rooms are concerned it is normally only intelligibility which is of interest, and quality is of secondary importance.

Colourations do occur in large auditoria but they can usually be traced to such causes as the mechanical vibration of structures. For instance, Liverpool Philharmonic Hall, which was investigated by the author and his colleagues in the B.B.C. Research Department (Somerville 1949) shows a marked colouration at a frequency of 110 c/s due to resonance of a series of identical wall sections constructed of plaster on metal lath. However, serious colourations of this kind are very exceptional in large auditoria, where the more usual defects are determined by time intervals rather than frequency variations, and take the form of echoes, unsuitable reverberation time, focussing effects and so forth.

Moreover, it is a curious fact that broadcasting organisations abroad attach little significance to colourations on speech programmes; the author remembers discussing the matter with acousticians of the Swiss P.T.T. with respect to one of their studios which had an intense colouration at about 150 c/s, yet which caused them little concern. In the B.B.C., much effort is devoted, particularly in new studios, in attempting to eliminate colourations by absorption and by careful choice of the positions of microphone and speaker. These measures are often partially, but seldom completely, successful.

During the past ten years the author has paid considerable attention to the phenomenon, publishing a contribution to the theory in a more general paper on small studios (Gilford 1959) and continuing the discussion in this present work which is bound with this thesis.

It is shown in the paper that colourations are due, fundamentally, to the fact that sounds of low frequency have wavelengths comparable with the dimensions of a small room. A sound-field set up in the room may be considered as forming a series of standing-wave systems with amplitudes dependent on the position of the source and the relationship between the wavelength and the dimensions of the room. Similarly, when a sound ceases, (such as for example a syllable of speech), the collapse of the direct sound and of that which has suffered a few reflections only is followed by the comparatively slow decay of the more highly excited normal modes and there is an audible change of pitch as this process takes place, due to the shift in the effective frequency. It is this shift in frequency which is considered to produce the subjective effect of colouration and an attempt is made in the paper to determine the conditions under which it will be audibly significant. However, though the theory there given explains many general observations, it has frequently been found to give inaccurate predictions. It is based on the assumption of a room in the form of a rectangular parrallelepiped. Some of the argument does not hold for non-parallel sided rooms, which nevertheless exhibit colourations, some at frequencies higher than those normally encountered in rectangular rooms of similar mean dimensions.

The success and failure of the theory there outlined may be illustrated by the following examples:-

<u>Example 1</u>

Dimensions: 13'8" x 7'11" x 8'0"

Fundamentals: 41 73 , 71 c/s

Isolated Groups of Modes at: 165, 212, 248, 286 c/s

Subjectively determined

colourations: 165, 210 c/s

Example 2

Dimensions:	20' x 13' x 9'3"			
Fundamentals:	28	43	61	c/s
Isolated Groups at:	86, 142, 170, 283 c/s			
Colourations at:	143 c/s only			

Example 3

Dimensions:	17' x 11' x 10'			
Fundamentals:	33	57	56	c/s
Isolated groups at:	100, 133, 200, 260 c/s			
Colourations at:	135, 260 c/s			

Example 4

Dimensions:	14.75' x 13.5' x 10.75'			
Fundamentals:	38	42	52	c/s
Isolated groups at:	81, 160, 190, 230 c/s			
Colourations at:	132 c/s			

Examination of these examples shows that the colouration frequencies usually coincided with isolated groups of modes, but there are in every case more isolated groups than observed colourations, and some colouration frequencies do not agree with the frequency of any isolated group.

Whatever the exact mechanism of the formation of colourations, it is to be expected that they may be reduced in severity by absorbing reverberant sound selectively either at the evident colouration frequency or possibly at one or more other frequencies. (This alternative must be considered as a possible necessity since a subjectively determined frequency may be a combination tone formed by two or more tones.)

Suppression of colourations by absorption has been attempted empirically by the writer. Unsatisfactory results may be due to several possible causes. The absorbers might be of too great a bandwidth so that an appreciable part of the spectrum is suppressed with the unwanted sound. Alternatively, the absorbers might be unfavourably placed in the room and may therefore not be effective at the particular frequency concerned. The arrangement and exact location of absorbers may have a great influence on their effectiveness as shown by Randall and Ward (1960). The relevant calculations on the influence of positions have been carried out by Wöhle (1956) for absorbers consisting of Helmholtz Resonators, and verified by him and by van Leeuven (1960). The case for absorbers with large frontal areas is much more complicated and has not previously been elucidated. It is, however, important in the present connection. The behaviour of such finite-area absorbers is also influenced by the effect of diffraction which causes an increase in the apparent absorption coefficient if the dimensions of the absorber are small or comparable with the wavelength of the sound. Some authors, as for example Kuhl (1960), have treated this phenomenon as an edge effect, which is presumed to vary in proportion to the length of the periphery of the absorber. The present author (1952-3), on the other hand, has treated it for low-frequency sound absorbers consisting of tuned membranes as a function of the radiation resistance of the absorber. This decreases as the absorber is reduced in size. The calculation of these effects is not yet on a sound scientific basis; if it were more complete it might be possible to build absorbing units in which a very high efficiency could be obtained at a specific frequency by correct construction and correct frontal area. This possibility is here investigated.

Finally, there is a need for better methods of assigning a frequency and magnitude to a colouration. The author has described methods of displaying isolated modes which in many rooms are associated with colourations but, as it has been remarked, no exact correlation with such modes has yet been established. A more reliable quasi-subjective method of classifying the colouration is required.

This present paper therefore sets out to examine the conditions for the formation of colourations at low frequencies in small rooms, to present new theoretical and experimental work on the behaviour of sound absorbing surfaces in small rooms and on the display and assessment of the colourations.

CHAPTER 2

THE NATURAL MODES OF ROOMS

2.1 Rectangular Rooms

Lord Rayleigh (1878) showed that a rectangular room has a triple infinity of natural modes given by the expression

$$f = \frac{c}{2}\sqrt{\frac{n_1^2}{\ell_1^2} + \frac{n_2^2}{\ell_2^2} + \frac{n_3^2}{\ell_3^2}} \text{ where } \ell_1, \ell_2, \ell_3 \quad \ldots (2.1)$$

are the dimensions of the room n_1, n_2, n_3 can have any integral values.

The modes represented by this equation fall into three main classes, as follows:-

Axial Modes

This term is used for those modes for which two out of the three n's are zero. These modes are associated with standing-wave patterns in which the particle velocities are all parallel to two of the three pairs of room boundaries.

Tangential Modes

These are all modes for which one only of the n's is zero. All particle velocities are then parallel to one pair of boundaries.

Oblique Modes

In these, all the n's have non-zero values and the particle velocities are oblique to all the surfaces.

It has been pointed out already that the number of modes in the audible range of frequency which are represented by this formula is extremely high. It may be shown (Bolt 1939) that the number lying below a frequency f is given approximately by:

$$\frac{4\pi}{3}\frac{f^3}{c^3}\cdot V - \frac{\pi f^2}{4c^2}\cdot S + \frac{f}{c}\;\frac{(\ell_1 + \ell_2 + \ell_3)}{2} \qquad \ldots\ldots \quad (2.2)$$

where $V = \ell_1\ell_2\ell_3 \qquad S = 2(\ell_1\ell_2 + \ell_2\ell_3 + \ell_3\ell_1)$

For a typical small room where $V = 100\ m^3$, $S = 120\ m^2$ and $\ell_1 + \ell_2 + \ell_3 = 15$ m we have 488 modes below 200 c/s.

It is clear that only a minute fraction of these are individually significant, and Mayo (1952) showed how their significance could be determined in the case of rectangular rooms.

Mayo examined the build-up and decay of sound pressure during and after the sounding of a tone by calculating the sums of the contributions of all the images of the source reflected in the walls of the room. This may be done comparatively easily for a rectangular room.

Fig. 2.1 (a) shows the array of images in one plane produced by a source S inside the room ABCD. Reflection in the four walls produces first-order images $S_{11}, S_{12}, S_{13}, S_{14}$. If the source is suddenly started, the sound will reach an observer O in the room first by the direct path SO and then along paths $SR_{11}O$, $SR_{12}O$, $SR_{13}O$, $SR_{14}O$ by reflection at the point R_1 on the walls. By simple geometry, $SR_{11} = S_{11}R_{11}$ and

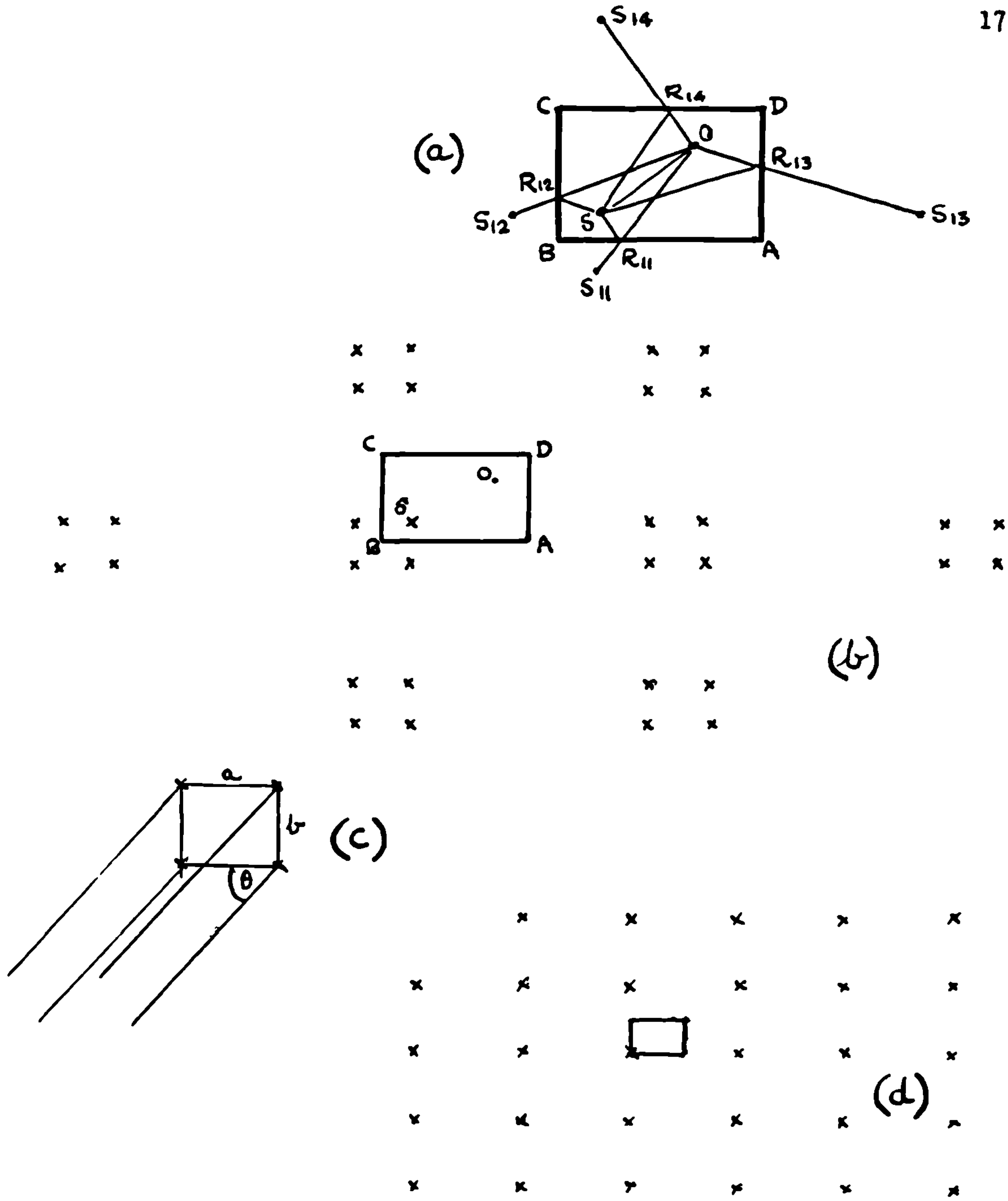


Fig. 2.1 Image Formation in a Rectangular Room

(a) Formation of Primary Images
(b) Clusters of Images formed by source at arbitrary position in Room
(c) Geometry of single Cluster
(d) Image Array from Source in Corner

therefore the time of arrival of the reflected sound from R_{11} is the same as if the sound had come from an image at S_{11} which appeared simultaneously with the starting of the source at S. The same applies to the other four images and by extension to the second-order images (i.e. images of the four images in the four walls) and generally to the images of the nth order.

Fig. 2.1 (b) shows the images of the first three orders. It will be noted that an image of order n can only be formed in a wall of the studio which is facing the image of order (n - 1) from which it is formed. Primary images will therefore give rise to three secondary images each, and images of the second and higher orders will generally have only two images.

It will be seen that the images are arranged in clusters of four corresponding to the configuration of the source and its nearest three images. When the source is started, therefore, the observer at O will receive an infinite series of groups of wave trains which will in general give an irregularly increasing sound field at O.

Consider in Fig. 2.1 (c) one such cluster which is at a distance so large that the angles between the lines joining O to the four images are small, i.e. that the lines are substantially parallel where they intersect the four images. Let the angle made by these lines with one pair of sides of the room be θ.

Then the distances of the four images from O are D, $D + a \cos\theta$, $D + b \sin\theta$, $D + a \cos\theta + b \sin\theta$, where D is the distance of the nearest of the four from O.

If the four images suddenly appear and we make the arrival of the first wavefront from the nearest image zero time, the first wavefronts from the others arrive at times $\frac{a\cos\theta}{c}$, $\frac{b\sin\theta}{c}$, $\frac{a\cos\theta + b\sin\theta}{c}$ respectively, where c is the velocity of sound.

The steady-state pressure due to the four images is given by

$$Po\left\{\sin\omega_o t + \sin\omega_o\left(t - \frac{b\sin\theta}{c}\right) + \sin\omega_o\left(t - \frac{a\cos\theta}{c}\right) + \sin\omega_o\left(t - \frac{a\cos\theta + b\sin\theta}{c}\right)\right\}$$

from an arbitrary zero time where the signal from the nearest image is zero.

Writing, $\frac{b\sin\theta}{c} = t_1$, $\frac{a\cos\theta}{c} = t_2$, the pressure is

$$Po\left\{\sin\omega_o t + \sin\omega_o(t - t_1) + \sin\omega_o(t - t_2) + \sin\omega_o(t - t_1 - t_2)\right\}$$

$$= Po\left\{\left[\sin\omega_o t + \sin\omega_o(t - t_1 - t_2)\right] + \left[\sin\omega_o(t - t_1) + \sin\omega_o(t - t_2)\right]\right\}$$

$$= 2Po\left\{\sin\tfrac{1}{2}\omega_o(2t - t_1 - t_2)\cos\frac{\omega_o}{2}(t_1 + t_2) + \sin\tfrac{1}{2}\omega_o(2t - t_1 - t_2)\cos\tfrac{1}{2}\omega_o(t_2 - t_1)\right\}$$

$$= 2Po\sin\omega_o\left(t - \frac{t_1 + t_2}{2}\right)\left\{\cos\frac{\omega_o}{2}(t_1 + t_2) + \cos\frac{\omega_o}{2}(t_2 - t_1)\right\} \quad (2.3)$$

Since t_1, t_2, ω_o are here constants, this represents an oscillation of frequency $\omega_o/2\pi$, lagging in phase by $\frac{\omega_o}{2\pi}(t_1 + t_2)$ behind the receipt of the signal from the nearest image.

The most important case is that in which the four images start simultaneously and the radiation from them with frequency $\omega_o/2\pi$ acts on a microphone or the ear of an observer.

The effect of a wave train from a single image has already been described, and it was shown that the radiation of the original frequency ω_o is replaced by a series of components with the frequencies of the natural modes of the room, those components nearest in frequency to ω_o having the greatest amplitude.

Mathematically, the excitation of any system at frequency ω is represented by the spectrum intensity at this frequency which is given by the Fourier transform of the signal.

Generally $S(\omega) = Re\int_{-\infty}^{+\infty} e^{-i\omega t} F(t).dt$ where $S(\omega)$ is the Fourier spectrum function and $F(t)$ is the time function of the signal. Re signifies that the real part of the integral only is taken.

If $F(t) = P_o \sin\omega_o t$, starting at time $t = 0$,

$$S(\omega) = P_o\, Re\int_0^{\infty} e^{-i\omega t}.\ \sin\omega_o t\ dt$$

Integrating by parts

$$\frac{S(\omega)}{P_o} = Re\left[\frac{-e^{-i\omega t}}{i\omega}\sin\omega_o t\right]_0^{\infty} + Re\int_0^{\infty}\frac{\omega_o}{i\omega}\cdot e^{-i\omega t}\cos\omega_o t.dt$$

$$= 0+Re\left\{\frac{\omega_o}{i\omega}\left[-\frac{e^{-i\omega t}}{i\omega}\cos\omega_o t\right]_0^{\infty} - \frac{\omega_o}{i\omega}\int_0^{\infty} e^{-i\omega t}\frac{\omega_o}{i\omega}\sin\omega_o t\ dt\right\}$$

$$= Re\left(0 + \frac{\omega_o}{i\omega}\cdot\frac{1}{i\omega} + \frac{\omega_o^2}{\omega^2}\int_0^{\infty} e^{-i\omega t}\sin\omega_o t\ dt\right)$$

$$= -\frac{\omega_o}{\omega^2} + \frac{\omega_o^2}{\omega^2}\cdot\frac{S(\omega)}{P_o}$$

$$\therefore \frac{S(\omega)}{P_o} = \frac{-1}{1-\omega_o^2/\omega^2}\cdot\frac{\omega_o}{\omega^2} = \frac{\omega_o}{\omega_o^2-\omega^2} \qquad \text{........ (2.4)}$$

The form of this spectrum is shown in Fig. 2.2.

The spectrum of the time function of the four images is

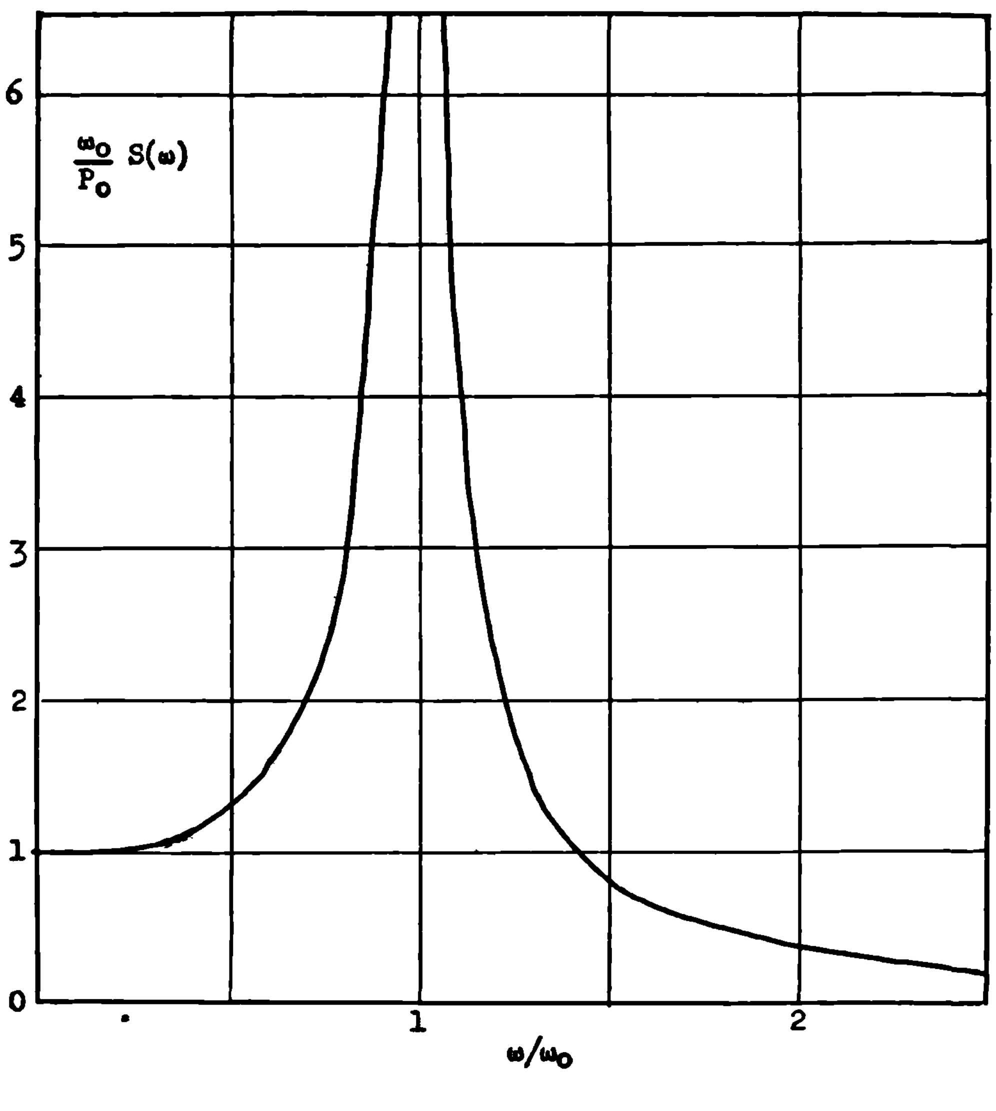


Fig. 2.2 Spectrum of train of waves $P_o \sin \omega_o t$ starting at $t = 0$

$$S(\omega) = P_o \left| \frac{\omega_o}{1 - (\omega/\omega_o)^2} \right|$$

the sum of terms

$$S(\omega)_n = \mathrm{Re}\int_{t_n}^{\infty} P_o\, e^{-i\omega t} \sin\omega_o(t - t_n)\, dt \qquad \ldots\ldots (2.5)$$

where t_n has the values o, t_1, t_2, $t_1 + t_2$.

Integrating by parts;

$$\frac{S(\omega)n}{P_o} = \mathrm{Re}\ \frac{-1}{1 - \frac{\omega_o^2}{\omega^2}} \cdot \frac{\omega}{\omega^2}\, e^{-i\omega t_n}$$

$$= \frac{\omega_o}{\omega_o^2 - \omega^2}\, e^{-i\omega t_n}$$

Hence the required spectrum is

$$S(\omega) = \mathrm{Re}\ \frac{P_o . \omega_o}{\omega_o^2 - \omega^2}\ (1 + e^{-i\omega t_1} + e^{-i\omega t_2} + e^{-i\omega(t_1 + t_2)}\} . (2.6$$

This represents the spectrum of a single waveform started at time 0 but modified by being multiplied by the sum of four vectors which rotate continuously about the origin as ω increases.

This produces a fluctuation about the simple spectrum of $P_o \sin\omega_o(t)$ but leaves the main maximum at $\omega = \omega_o$ unchanged.

These fluctuations represent the effect on the excitation of the position of the source in the room since they depend on the magnitude of a, b and θ. They modify the magnitude and phase of the response of a microphone to the transients when the source is started or stopped, in comparison with the behaviour when the source is in a corner, therefore giving rise to coincident images. The main feature of the spectrum, however, the infinity at $\omega = \omega_0$, remains unchanged.

Hence we may regard the behaviour of the source and its three images as essentially equivalent in its main effects to that of a single source.

Thus we may, without loss of generality, choose the source to be at one corner, say B in Fig. 2.1 (a), of the room so that we have a single image in place of the four. This substitution of a single image facilitates the theoretical work, the image diagram then reducing to that shown in Fig. 2.1 (d).

Mayo (1952) shows that three classes of image spacing are possible, giving rise to three types of characteristic frequency. First, we have short-path reflections arriving at virtually random intervals from the nearer images. There is no order in the time-spacing of these reflections because the distances from the observer in terms of the image spacings are irrational. Second reflections of regular spacing begin to arrive from directions along lines of images passing through or near to the room, giving rise to what Mayo terms "line frequencies" and their harmonics. These are established early on in the period following the switching on of the source and die away rapidly after the source is switched off because the amplitude of the reflection from a distant image is attenuated partly by the inverse-distance law and by the loss by absorption in the n reflections it has undergone.

Lastly, we have the true modal frequencies as defined at the beginning of this section, produced by equally-spaced planes of images. These are late in forming and in dying away because only planes at a considerable distance from the observer give sufficiently plane wavefronts to produce regular reflection intervals. They die away slowly after switching off the source because, being plane waves, they have no inverse-distance attenuation.

Mayo shows that only those frequencies which are common to both the line arrays and plane arrays of images will reach a great enough initial amplitude and decay slowly enough to be audibly significant. In the case of a rectangular room this condition is fulfilled only by the frequencies corresponding to axial modes of the room, i.e. where only one out of n_1, n_2, n_3 of the expression for the natural modal frequencies given above is non-zero. This condition corresponds to stationary wave systems parallel to one or other of the pairs of room surfaces.

The author has shown (1959) that for a mode to be audible as colouration , moreover, a separation of the order of 20 c/s from the adjacent modes in the frequency scale is necessary. Such separations occur for a few modes in most rooms of the size of a typical living room and cannot in general be avoided.

These two considerations enable a prediction to be made of the colourations in rectangular rooms. For reasons explained in the same paper, they seldom occur in acoustically treated rooms above about 300 c/s, and it is usual for between two and four colourations to be indicated below this frequency. The agreement with prediction is not, however, by any means complete.

2.2 Non-Rectangular Rooms

The image pattern concept is not applicable to non-rectangular rooms because even slight non-parallelism introduces great complications into the pattern and also upsets the image redundancy which greatly simplifies the rectangular image pattern. Fig. 2.3 shows the first three orders of images for a rectangular room with one pair of walls only, making an angle of 5° to each other.

Nevertheless, one is inclined to suppose that a perturbation of this order would not make any striking difference to the frequencies of the principal modes, at any rate of the lower order modes, and some success has been obtained in the past in predicting colouration frequencies of nearly-rectangular rooms by regarding them as rectangular with dimensions equal to the mean of the length of opposite sides. However, it is clear that for large deviations from rectangular shape the assumption will eventually cease to be valid, and, indeed, it ceases to have any theoretical validity even at very small deviations if the image-space does not have a simple form.

It was therefore decided to conduct experiments to find out whether the results of the image-space approach remained true even in some modified form as the shape of the room deviated from a true rectangular shape.

An investigation in three dimensions would be very

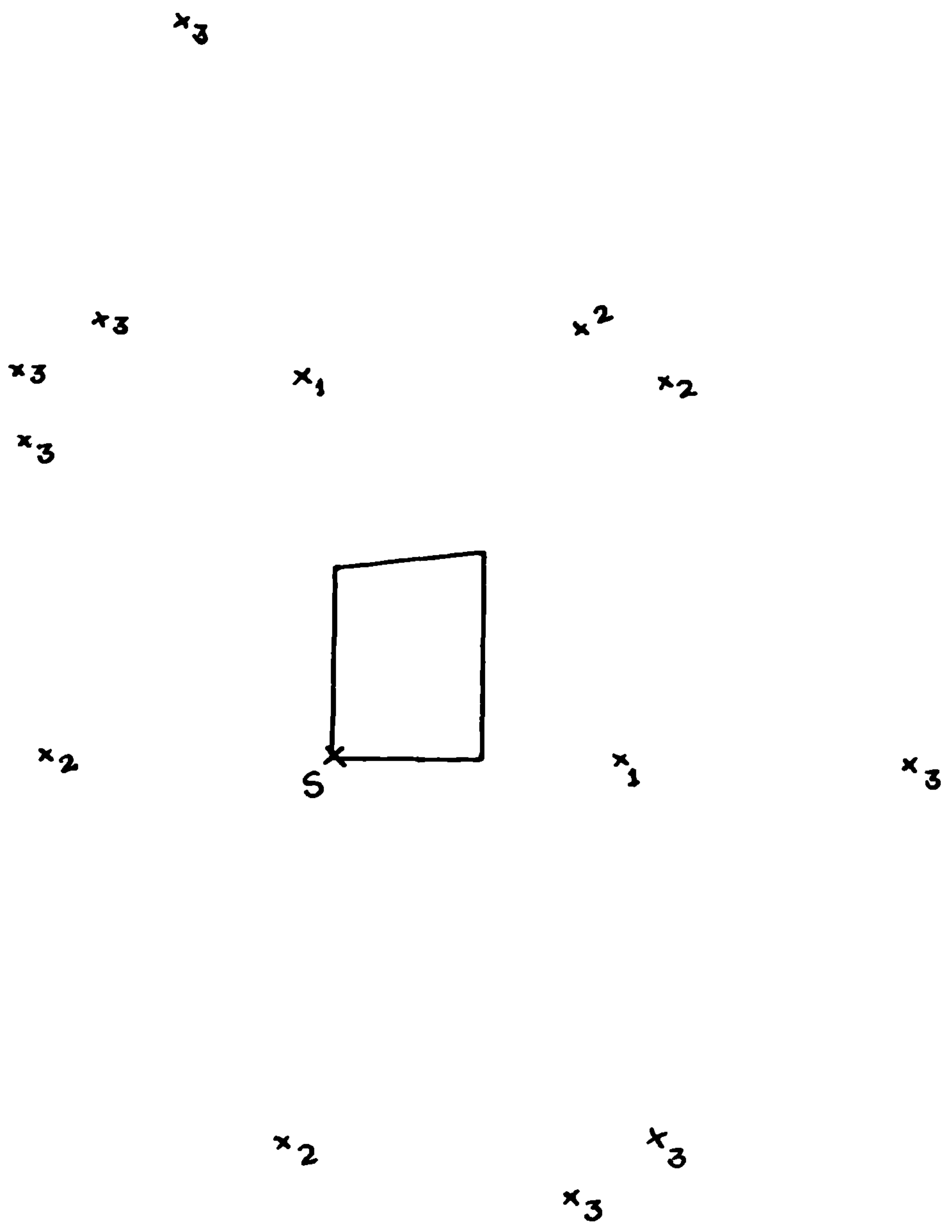


Fig. 2.3 First Three Orders of Images of a Source (S) in one Corner of a Room with one Skew Side.

The Numbers by the Images indicate their Order complicated since it would entail the construction of rooms of unusual shape. Moreover, the rapid closing of the frequency gaps between normal modes constitutes a difficulty since only at low frequencies would individual modes be separately distinguishable.

To simplify the experimental work and the interpretation of results, it was decided to carry out the experiments in model spaces, with frequencies scaled up in the inverse ratio of the dimensions of the model to that of a typical small room, but to make one dimension very much smaller than the other two. A model size approximately 90 cm x 60 cm was taken as being of a convenient size, and these dimensions are about one fifth those of an ordinary small room. The frequency range of interest is therefore from approximately 250 c/s to 1500 c/s.

The third dimension was fixed at 5 cm for which the lowest natural frequency is 3600 c/s, well outside the region of interest.

Fig. 2.4 shows diagrams of three models built for the experiments. Model A was rectangular with internal dimensions 74.4 cm x 49.5 cm x 5 cm. Model B had one short side at an angle of 5° from the opposite side but the dimensions are such that the mean of the two long sides was equal to the larger internal dimension of model A. Model C had two opposite right angles, the other two angles being supplementary and differing by 10°. The mean dimensions along and across the model were equal to the sides of model A. Each model had a frame constructed of timber 5 cm x 5 cm square section. The bottom surface was formed by a

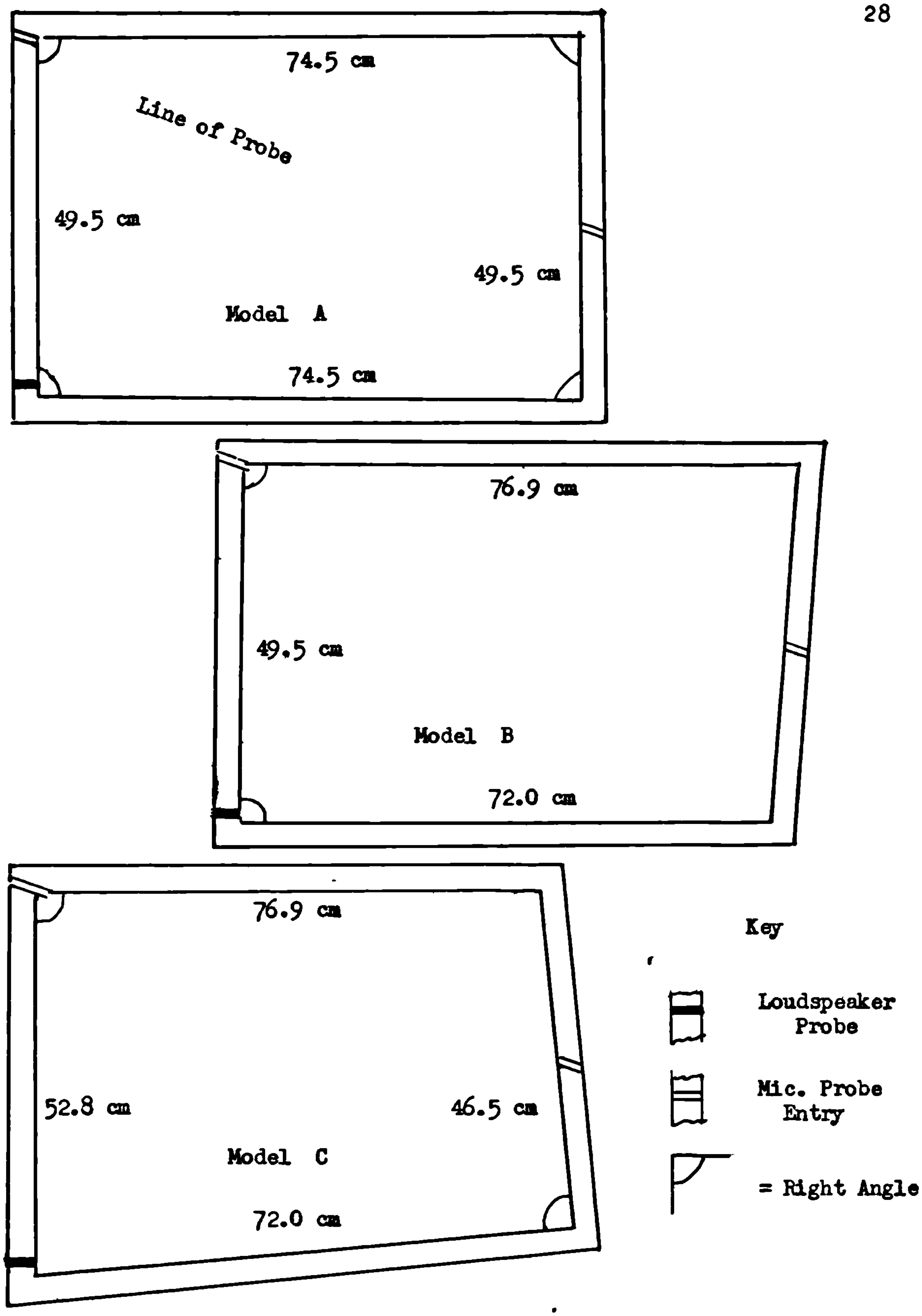


Fig. 2.4 Diagrams of the Three Flat Models

smooth floor tiled with a hard asbestos /P.V.C. mixture and the top surface by a piece of plate glass 6 mm thick. The edges of the frame were sealed to the floor and the plate glass by thin strips of soft wax and the interior surfaces of the wooden frame were smoothed with glasspaper and finished with two coats of paint, the second coat having a hard glossy surface. A hole was bored near one corner of each frame in the position indicated in the figure to permit the insertion of a tube of 12 mm internal diameter connected to a small loudspeaker unit of the type used to drive cellular-horn high-frequency "tweeters". The connecting tube was initially packed with pieces of sleeving of approximately 1 mm internal diameter for damping its organ-pipe resonances, but this was later removed as it proved to be unnecessary and reduced the efficiency of the loudspeaker. A second hole was made in the short side furthest from the loudspeaker to admit a probe tube connected to a moving-coil microphone. The hole emerged in the centre of the side and was drilled along a line connecting this point to the corner nearest to that occupied by the loudspeaker tube. The microphone was a Sennheiser MD3, designed for use with a probe tube and a special probe was fitted, 76 cm long and 5 mm internal diameter. Initially this was damped with a fine sliver of cellulose acetate wadding.

The placing of the microphone probe hole was designed to allow several significant positions in the model to be studied by sliding the tube to different distances. These were:

(A) Corner positions where all modes of air space should be fully represented with equal excitation.

(E) A position halfway along one side by which the fundamental, and all odd multiples of the fundamental resonance frequency in one dimension, could be suppressed.

(D) A position halfway between the microphone probe hole and the far corner at which the pressure of the fundamental in the other dimension and its odd harmonics should be zero.

(B), (C) Positions one-third and two-thirds the distance between (A) and (E).

It was intended to make preliminary experiments with the rectangular model, identifying the modes, determining their class and rates of damping and then to compare the results with those from the two perturbed rectangles.

The microphone-loudspeaker combination was first tested to determine its frequency characteristic. In initial tests the damping was present in both the loudspeaker and microphone tubes but the ratio of wanted signal to noise was extremely small. The microphone tube was then sealed with a small piece of Plasticine and it was found that the transmission was little affected. Removal of all the wadding from the microphone tube increased the signal-to-noise ratio by 30 dB. The frequency characteristic was then as shown in Fig. 2.5 (a). Subsequent removal of the sleeving from the loudspeaker tube produced the second curve, Fig. 2.5 (b),

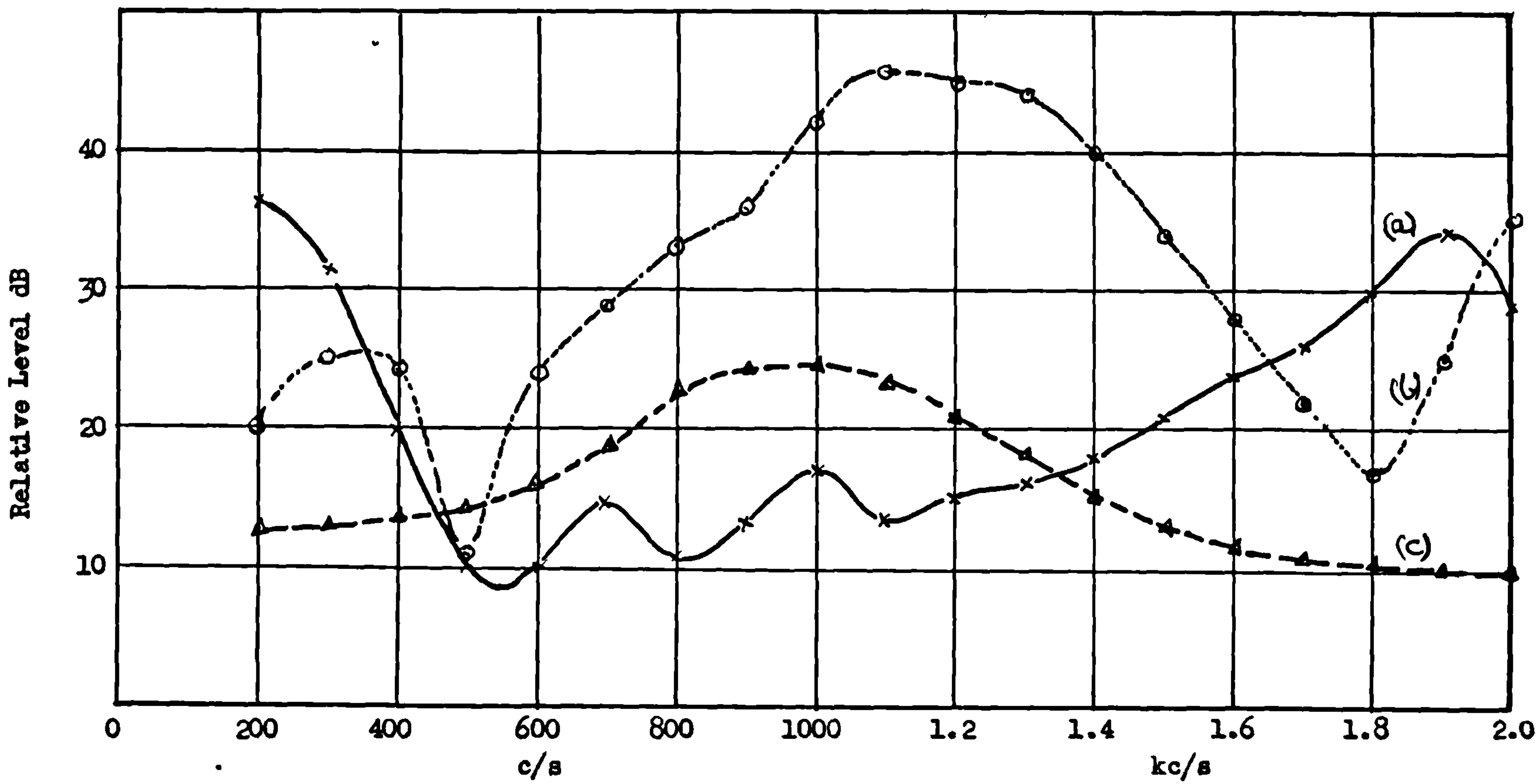


Fig. 2.5 Steady State Frequency Characteristic of Loudspeaker-Microphone Combination

(a) With damping in loudspeaker tube
(b) Without " " " "
(c) With intermediate quantities of damping in loudspeaker and microphone tubes

which shows a resonance at 1100 c/s associated with the tube length. Fig. 2.5 (c) shows a compromise condition, finally adopted, using small amounts of damping material in the tubes.

The frequency characteristics of the combination in free space conditions and in the models were recorded by means of a Bruel and Kjaer level recorder, using gliding tone from a B.B.C. oscillator (Mayo, Beadle et al 1951).

Fig. 2.6 shows the steady-state curves for the rectangular model in three different positions of the probe tube. It will be seen that all three are characterised by marked peaks and minima of pressure, the peaks being according to theory at the modal frequencies of the system.

The following table gives a list of the calculated modal frequencies of the rectangular frame.

TABLE 2.1

Calculated Frequencies of
(c/s)
Rectangular Model A.

$$f = \frac{c}{2}\sqrt{\left(\frac{n_1}{\ell_1}\right)^2 + \left(\frac{n_2}{\ell_2}\right)^2}$$

$n_1, n_2 = 0, 1, 2, 3 \ldots\ldots\ldots$

$\ell_1 = 74.4$ cm.

$\ell_2 = 49.5$ cm.

$n_2 \downarrow$ / $n_1 \rightarrow$	0	1	2	3	4	5	6	7	8
0		230	460	690	921	1151	1381	1611	1842
1	345	415	576	772	984	1202	1423	1649	1870
2	690	728	830	977	1151	1343	1545	1754	1968
3	1036	1062	1134	1246	1387	1550	1727	1916	
4	1381	1401	1457	1545	1661	1797	1954		
5	1726	1743	1788	1860	1956				

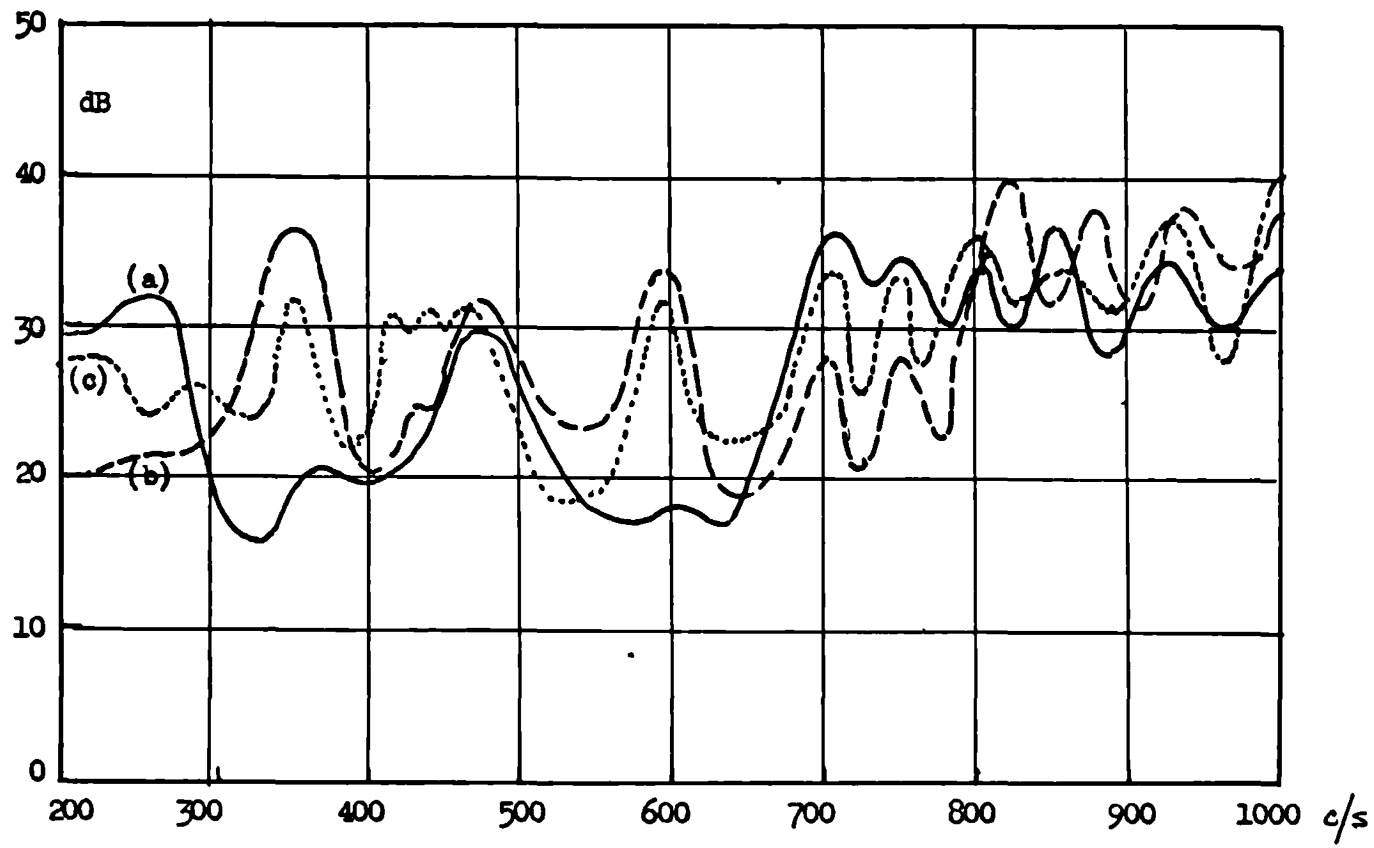


Fig. 2.6 Steady-state Curves of Rectangular Model ('A') for three Positions of Probe

(a) Near Entry at Corner (b) $\frac{3}{4}$-way across the model

(c) Near Entry in Side

The measured peak-pressure frequencies are compared with these calculated frequencies in Fig. 2.7.

It will be seen from the figure that

(1) All the axial mode frequencies appear, but they are in general, particularly the low-frequency modes, displaced towards a higher frequency.

(2) Only a few of the tangential frequencies are clearly represented, but there is a number of maxima additional to the axial modes very similar to the number of calculated non-axial modes, as shown in the following table:

TABLE 2.2

Measured and Calculated Numbers of Non-Axial Modes in Flat Model

Frequency Interval	No. of Calculated Non-Axial Modes	No. of Measured other than Axial Modes
0 - 500	1	1
500 - 1000	6	5
1000 - 1500	8	4
1500 - 2000	7	4

The low figure of measured modes for the octave 1000 - 2000 should be supplemented by a small number of subsidiary peaks which were observed but were neglected as it was difficult to assign frequencies to them.

The first anomaly, the shift towards higher frequencies, was first considered. A shift in this direction is unlikely to

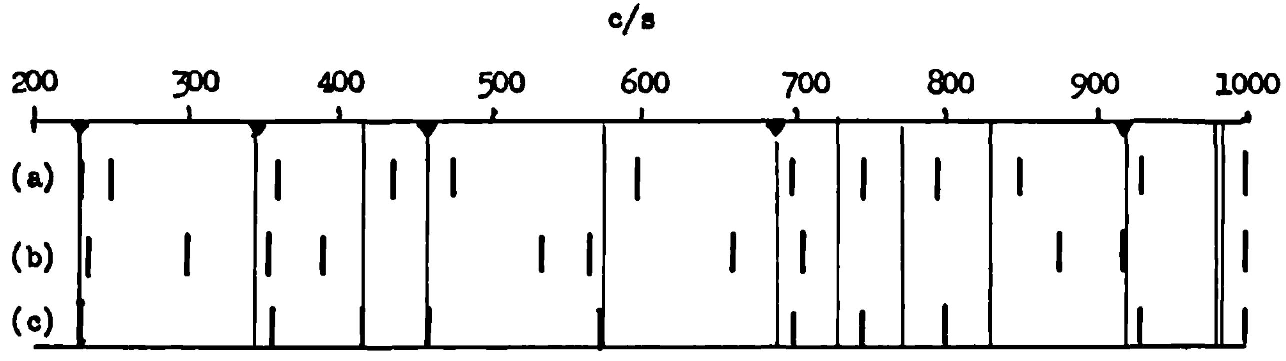


Fig. 2.7 Agreement between Measured Peak Frequencies and Calculated Mode Frequencies

Thin continuous line	-	Mode Frequencies Axial Modes marked thus ▼
Short lines	-	Measured Peak Frequencies (Three positions)

be caused by compliance in the sides of the model, which would be expected to produce a downward shift in frequency. Nevertheless, it was thought worthwhile to test the effect of compliance in one face of the model, by replacing the glass top of the model by a sheet of 20 s.w.g. aluminium. Two measurements of the peak frequencies were made, one with the aluminium sheet alone, the other with the sheet held down round the edges by a frame similar to the model frame secured by weights above.

The results are shown in the following table in which the mean frequency of the peaks corresponding to the axial frequency are compared with the values previously obtained with the glass top and then calculated for axial modes.

TABLE 2.3

Effect of Compliance in Top Face of Model

Frequencies of Maxima (c/s)

Aluminium Top	Glass Top	Calculated
250	250	231
360	364	345
474	460	460
700	704	690
930	925	920
1050	1055	1040
1155	1160	1155
1405	1400	1387
1620	1620	1617

Contd.

TABLE 2.3 (Contd.)

Aluminium Top	Glass Top	Calculated
1750	1750	1725
1845		1848

There is no further systematic shift towards higher frequencies in these results and it is therefore concluded that the shift in the first experiment is not due to compliance.

As a further check of the method and the accuracy of the equipment, experiments were carried out in a pipe of comparable length to the longer dimension of the model, the exact dimensions being: length 102.1 cm, diameter 10.0 cm. The modal frequencies of a pipe are, of course, easily calculated and the end corrections accurately known.

The following table shows the results.

TABLE 2.4

Calculated and Measured Modal Frequencies of Pipe.

Mode No.	Calculated Modal Frequency	Observed Frequency of Maximum Excitation	Difference %
1	168	170.5	1.50
2	336	342	1.98
3	503	511	1.60
4	690	680	1.49
5	838	848	1.18
6	1005	1017	1.20
7	1187	1190	0.50
8	1340	1352	0.92
9	1505	1517	0.80
10	1675	1690	0.89
11	1840	1860	1.08
12	2010	2010	0.00
		Mean difference =	1.08

This mean percentage difference between calculated and measured frequencies is small compared with those obtained from the flat model. The systematic discrepancies for the flat model remain unexplained.

The second point, the failure to identify some of the measured modal frequencies with any of the calculated axial or tangential modes was investigated by making sound pressure measurements over the whole area of the model. The position of the loudspeaker aperture was left unchanged but the microphone probe entry was modified to allow the probe to be turned obliquely from its normal direction. To do this, an aluminium plate which had been drilled to receive a rubber grommet just large enough to fit the probe tube was screwed inside the frame to cover the probe entry, and the existing hole through the wood was widened laterally to allow the probe tube to move freely behind the grommet. In this way, the tip could reach any point in the further half of the model and about half of the area of the nearer half.

Paper marked out with a square grid of lines was laid in the bottom of the model to provide a co-ordinate system for specifying the position of the probe entry. The microphone output after amplification was fed to an amplifier-detector, thus enabling the relative pressure to be measured on a decibel scale.

The probe was inserted into a further corner of the model and the oscillator was adjusted to one of the frequencies of maximum response.

The amplifier-detector was adjusted to give a scale reading near to the maximum and then the probe was swept across the area of the model to locate the regions of maximum and minimum pressure. The pressure at a point (x, y) is given by

$$p = P(\cos\frac{\pi n_x x}{\ell_x}\cos\frac{\pi n_y y}{\ell_y}$$ where ℓ_x, ℓ_y are the

dimensions of the rectangle and n_x, n_y are integers.

The equation defines antinodal lines dividing the area equally into n_x and n_y zones in the two directions, each zone being a rectangle of dimension ℓ_x/n_x and ℓ_y/n_y.

Where two modes are excited at the same frequency, the two pressure fields will be superimposed, causing a complex disturbance of pressure.

By examination of the fields, the following modes were identified.

TABLE 2.5

Identification of Modes.

Mode	Actual Frequency c/s	Calculated Frequency c/s
(1,0,0)	255	230
(0,1,0)	360	345
(1,1,0)	474	460
(2,0,0)	450	420
(2,1,0)	592	576
(0,2,0)	700	690
(0,3,0)	745	728
(3,1,0)	795	772
(2,2,0)	850	830
(4;0,0)	930	921
(4,1,0)	995	984
(3,2,0)	1000	980
(1,3,0)	1060	1062
(5,1,0)	1220	1210
(3,3,0)	1260	1250
(5,2,0)	1350	1355
(6,0,0)	1400	1380
(6,1,0)	1445	1430
(3,4,0)	1555	1570
(7,0,0)	1620	1610

This shows that, allowing for a nearly constant difference between calculated and measured frequencies, there is a one-to-one correspondence between the observed maxima and the calculated axial or tangential modes.

The modal frequencies of the three models are summarised in the table below. In this table, the full list of all the identified modes of the rectangular model "A" (See Fig. 2.4) appears on the left hand side, while the frequencies at which maxima appear in the non-rectangular models in positions at which the particular mode would be expected to be detected are given on the same horizontal lines.

TABLE 2.6

Observed Modal Frequencies in the Three Models.

Model A	Model B	Model C
255	250	245
360	362	370
450	437	450
474	474	480
592	593	597
700	700	700
745	745	760
795	790	798
850	840	828
930	927	927
1000	1000	1000
1060	1050	1030
1155	1150	1155
1220	1220	1230
1260	1260	1260
1350	1345	1350
1400	1400	-
1445	1445	1450
1555	1555	1520
1620	1620	-

At low frequencies all the shapes show the same frequencies without omissions or systematic differences. At high frequencies a remarkable similarity persists, but there are a few omissions or large differences.

It is concluded that non-rectangular rooms with moderate angles of divergence may be regarded, from the standpoint of modal frequency distribution, as rectangular rooms with the same mean dimensions. If external circumstances make the use of a non-rectangular space obligatory, prediction of possible colourations should be carried out in the same way as for rectangular rooms.

Schubert and Steffen (1961) carried out scale model experiments on the effect of changes in the angle of one wall of a concrete enclosure which was rectangular in its original condition. The movable wall was hinged along one edge, so that the mean dimensions of the enclosure in the direction of movement of the other end was reduced by one half of that movement. Following the author, (1959) they assumed that the wide spacing of the modes was the cause of colourations, and thus confined their observations to the low-frequency region within approximately two octaves of the lowest mode frequency. However, unlike the author, they included non-axial eigentones in their considerations.

The frequencies of all modes were measured and plotted as a line spectrum for each of fifteen angles of the movable wall. For every frequency the mode was identified, presumably by methods similar to those described above.

Their conclusions were:

(1) That as the angle of the movable wall was increased, the effective reflecting plane was shifted inwards, thereby raising the frequencies of all modes in which this surface participated. This shift is not equivalent to the mean displacement of the surface, as might be expected, but appears to be equal, with long-wavelength sound, to the linear movement of the displaced end of the wall, i.e. twice the mean displacement of the wall. Hence, the modes separate even if the mean length of the room is maintained constant.

(2) If the inclination of the wall reaches 15°, the lowest mode in that direction ceases to rise in frequency, but transverse modes start to change in frequency.

(3) The eigenfrequency density decreases with increasing angle of inclination. Making corrections for the reduced volume, the number of modes below 450 c/s was reduced to two thirds when the inclination was increased sufficiently to transform the rectangle into a triangle.

TABLE 2.7

Comparison of Rectangular and Non-Rectangular Spaces.
(After Schubert and Steffen)

	Number of discrete modes (Qn) 450 c/s	Mean-square Separation of modes (δ)	(Qn/δ)
Rectangle	16	9.14×10^{-2}	175.3
Trapezium 7°	12	11.11	108
Trapezium 12°	11	12.74	86.4

Table 2.7 shows the evident superiority of the rectangular room with respect to eigenfrequency density.

The first column shows the total number of eigenfrequencies below 450 c/s, while the second is a spacing index defined by the expression

$$\delta = \sqrt{\frac{\sum_{f_a}^{f_b}(f_{n+1}-f_n)^2}{n\, f_a\, f_b}}$$

where f_a is the lowest frequency, f_b the next above 450 c/s and n the total number of intervals. This expression gives the mean square spacing, normalised by dividing by the geometric mean of the frequency range.

These results together with the assumption quoted above that the occurrence of colourations is associated with large spacings, rather than agglomerations of separated modes, agrees with the subjective observation that rectangular rooms give better speech quality than non-rectangular rooms of equal volume.

There are two points of difference between this result and the published findings of the author.

(1) In Schubert and Steffen's work, there was a greater apparent change in the frequencies of modes involving the movable wall than in the author's experiments. This could be due to the fact that in the author's experiments the total area of the model was maintained constant between the two wall configurations, whereas Schubert and Steffen's model changed in area since the

movable wall was hinged along one edge. If this is so, the shift in the apparent position of the reflecting plane should be one half that of one end of the reflecting plane, the other being a hinge point. This should hold according to Skudrzyk (1954) up to a value of d/λ equal to 0.3 where d is the displacement of the moving end of the wall, after which the apparent shift of the reflecting plane should rise slowly. These results however start with a very high initial shift for small d/λ almost equal to the displacement of the moving end, drop to 0.4 at $d/\lambda = 0.3$ and fall thereafter to a minimum of 0.15 at $d/\lambda = 0.45$.

The author's own work did not however support this change, being exactly in agreement with Skudrzyk's theoretical curve at d/λ up to over 0.2. For higher values at the fixed wall angle of 5° the modes became difficult to identify and a side of greater obliquity would be desirable to check the point.

(2) Schubert and Steffen considered tangential as well as axial modes. This has implications for the rectangular mode theory dealt with elsewhere.

(3) On the subjective side, the two authors find that colourations are of more significance in listening rooms than in studios, since, they argue, it is both usual and possible to use microphones at smaller distances in a studio than the loudspeaker-listener distance in the listening room or cubicle. The present author finds that colourations are much more serious in the studio than the listening room because those generated in the listening room

are partially rejected by the binaural image-separation process. Microphone distances in the studio are usually great enough for the reverberant sound to be easily audible over a transmission chain and with a monaural system it is indistinguishable from the direct sound.

The ratio of the reverberant sound to the direct at distance r is found as follows:-

Suppose the power radiated by the source is W; the intensity at distance r is then given by

$$I_d = \frac{W}{4\pi r^2}$$

The power supplied to the reverberant sound field is the total power from the source diminished by that associated with the direct sound. This latter component, in the steady state, is equal to the absorption by the room surfaces at the first reflection

i.e. $W_D = W\bar{\alpha}$, $\bar{\alpha}$ = mean coefficient of absorption of surfaces

Thus the power supplied to the reverberant field for all modes within a specified frequency range is

$$W_R = W(1 - \bar{\alpha})$$

Let the reverberant intensity be I_R.

The energy density is then I_R/c.

The mean free path of sound between reflections at the room surfaces is 4V/S (Kosten 1960) and hence the number of reflections in unit time is $\frac{Sc}{4V}$.

(Where V = volume of room. S = total area of surfaces.)

Hence the rate of dissipation of the reverberant sound energy is given by

$$W_R = W(1 - \bar{\alpha}) = V.\frac{I_R}{c} \cdot \frac{Sc}{4V} \cdot \bar{\alpha}$$

whence

$$I_R = \frac{4W(1 - \bar{\alpha})}{S\bar{\alpha}}$$

Therefore $$\frac{I_R}{I_d} = \frac{16\pi r^2 (1 - \bar{\alpha})}{\bar{\alpha} S} \qquad \ldots\ldots (2.7)$$

This is correct for an omnidirectional source. If the source has a directivity factor χ (defined as the intensity in specified direction divided by mean polar intensity) the ratio must be diminished by the factor χ giving

$$\frac{I_R}{} = \frac{16\pi r^2 (1 - \bar{\alpha})}{\chi \bar{\alpha} s} \qquad \ldots\ldots (2.8)$$

Consider a studio and a listening room of equal surface area and mean absorption coefficient. Assume that the studio microphone (a pressure gradient type with $\chi = \frac{1}{3}$) is 45 cm from a a speaker and that the listener is 120 cm from the loudspeaker in the listening room. At the low frequencies considered χ for the loudspeaker is approximately unity. For $S = 60\ m^2$ and $\bar{\alpha} = 0.3$, then, for the studio $\frac{I_R}{I_d} = 0.14$ and for the listening room $\frac{I_R}{I_d} = 3.00$.

The ratio between these two quantities is $\frac{3.00}{0.14} = 21.5$, or 13 dB which bears out the statement of Schubert and Steffen.

Binaural listening per se certainly gives a reduced sensitivity to reverberant sound. A rough comparison can be made in two ways: by covering one ear when listening to sounds in the room itself, or by comparing stereophonic and monophonic transmissions by loudspeaker. Neither of these methods shows differences equivalent to differences greater than about 3 to 6 dB in direct/reverberant intensity ratios. (This is a subjective assessment by the author and of several colleagues who have been professionally concerned with stereophonic reproduction.)

Comparison of a monophonic transmission from a loudspeaker with the sound field in the listening room, however, introduces psychological factors in addition.

(1) Increased attention to the loudspeaker as a localised source of interest which may lead to the virtual exclusion of information from other sources in the listening room. (This effect has recently been described by Cherry (1963).

(2) The establishment of the acoustics of the listening room as a norm, on which the reverberant components of the incoming programmes are superimposed.

Considerations such as these must be invoked to explain the ability of a listener to detect colourations in an electrically coupled room in the presence of the colourations of the listening room and to give them overriding subjective importance. The existence of this ability is well known to all who are concerned with the judgment of quality of broadcast transmissions.

Some experiments by the author and W.K.E. Geddes (1954) verified the effect. The experiments were designed to determine the best acoustic environment for listening, not to verify or disprove the value of listening in a reverberant environment. Speech from six different talks studios were judged by listeners in each of five acoustic environments. The majority of the subjects, but not all, were skilled in quality appraisal. Judgments of preference were made in each environment between each of the six studios paired in all fifteen combinations. Thus, a studio "A" was compared with Studio "B" and other studios represented by "C" were compared with both "A" and "B". If a subject submitted answers

$A > B$, $B > C$, $A > C$ ($>$ = better than)

this could be regarded as a consistent triad.

However, $A > B$, $B > C$, $C > A$ would be an inconsistent triad.

The mean number of inconsistent triads in the 15 judgments by all observers was used as an indication of the difficulty in judging differences in speech quality in that particular acoustic environment. The maximum number of inconsistent triads was eight, and the following table shows the actual mean proportion of the maximum.

TABLE 2.8

Inconsistent Triads in the Listening Rooms.

Listening Room	Inconsistent Triads divided by Max. possible number
a	0.07
b	0.17
c	0.18
d	0.11
a (repeat)	0.06

Room a was an acoustically treated listening room with total absorption rather less than that assumed in the numerical example above. The degree of consistency (only one inconsistent triad out of approximately 14 possible) shows a high degree of ability by the subjects to judge the reverberant sound of the studio in the presence of a much stronger reverberant field in the room.

Room d was a room with very low reverberation time. Rooms b and c had very much higher reverberation times than room a but even in these rooms the mean number of inconsistent triads was less than one sixth of the maximum number.

Reverting to the general question of the behaviour of non-parallel walled rooms, we conclude that, although image-array methods cannot be used, and it is therefore not valid to assume that it is only axial modes that are audibly significant, there are marked similarities between rectangular rooms and those with non-parallel wall pairs. There is some experimental evidence that simple tangential modes may be more important in non-rectangular rooms than in rectangular rooms, causing marked prolongation of reverberation at certain frequencied by reflection of sound between patches of reflecting surface on the four walls. This evidence, which is based on partially successful attempts to suppress strong colourations in two non-rectangular studios by means of local absorption is too scanty to merit presentation here.

In the absence of better methods of dealing with non-rectangular rooms, therefore, it is preferable to design small rooms of rectangular shape because, as shown by the evidence presented in this section, not only is irregular spacing of modes less probable, but the knowledge of mode spacing and relative importance is known with more certainty and the results can be predicted with greater chance of success.

2.3 Effect of Perturbations of the Walls

The foregoing sections have dealt with the normal modes of rooms having plane uniformly surfaced walls. It is important to consider also the effects of surface irregularities. Such irregularities are often introduced with the object of improving the diffusion of the sound field and though small in comparison with the dimensions of the surface are of appreciable size compared with the wavelength of sound. The effect of such irregularities has been investigated mathematically by Head (1953) who examined the case of a room in which one wall has a number of equally spaced identical projections.

Head examines an effectively two-dimensional case of a room in which one dimension, the z-dimension, is very small compared with the other two. It is bounded by two planes perpendicular to the z-axis and close together, and four walls perpendicular to these planes, these walls being plane and the fourth having a number of equally shaped and spaced projections.

These projections are part-cylinders or prisms with their generators parallel to the z-axis, their sections being part-circular, rectangular and triangular respectively.

Head uses a method devised by Feshbach (1944) to derive the eigenfrequencies associated with the different forms of projection, and shows that the presence of the projections causes a shift in the frequency of each natural mode of the unperturbed room. These shifts in frequency are inversely proportional to the difference of the squares of the plane-wall eigenfrequency and the nearest neighbouring frequency, and are therefore greatest for closely spaced groups of modes. A single very prominent mode will be replaced by a linear combination of neighbouring eigentone potentials and will then be less prominent. The mean frequency of maximum response of the room, however, will be substantially unaltered.

Rectangular projections are more effective in producing this result than are projections of semicircular or triangular section. Head shows that this result is due to the fact that

sections of
the rectangular projections have finite portions perpendicular to the wall on which they are based, whereas circular section projections have only infinitesimal perpendicular portions and triangular-section projections none at all.

We conclude that it is unnecessary to take into account the presence of projections or obliquities in the sides of an enclosure in calculating the modal frequencies and that the methods given previously in this section for estimating the most probable colouration frequencies are applicable for both oblique and perturbed-boundary rooms, so long as the mean dimensions are used in the calculations.

The selective treatment of prominent modes or groups of modes with nearly coincident frequencies is, however, of subjective importance and this will be dealt with again in a later section on diffusion.

2.4 Interaction between Neighbouring Modes

The instrumental methods of displaying colourations, to be described later, (Chapter 4), depend in some instances on the recognition of cathode-ray oscilloscope decay traces characteristic of frequency regions with strong isolated modes.

When tone of specified frequency is radiated into an enclosure, neighbouring modes in the room will be excited to an amplitude depending upon the sound pressure, the position of the sound source, the frequency separation of the signal from the mode and the bandwidth of the mode. The pressure/time function will vary from place to place in the room according to the standing-wave patterns associated with the modes. We will now see to what extent interaction between sets of neighbouring modes gives rise to recognisable decay functions displayed as time-functions of the pressure at a point in the room.

Fig. 2.8 represents a frequency scale with several modal frequencies $\frac{\omega_1}{2\pi}, \frac{\omega_2}{2\pi} \ldots \frac{\omega_r}{2\pi}$ marked upon it. Pure tone of frequency $\frac{\omega}{2\pi}$ is radiated into the room at constant power and excites the modes ω_1 to ω_5 to varying degrees.

The excitation will depend on

(1) The frequencies ω and ω_r of the driving tone and the mode.

(2) The acoustic resistance of the mode R_r which may be shown (e.g. Van Leeuwen 1960) for an axial mode to be

$$R_r = \frac{\pi \rho c^2}{6.9V} \quad \ldots\ldots\ldots\ldots (2.9)$$

R_r = R.T. of mode

V = volume of room

ρ = density of air

c = velocity of sound

(3) The characteristic function of the mode at the position of the source ψ_r, i.e. the pressure at the source expressed as a fraction of an antinode of pressure.

By analogy with a fixed mechanical oscillation, treatment of which will be found in text books on acoustics or vibration (e.g. Davis 1932), the excitation of the rth mode will be seen to be:

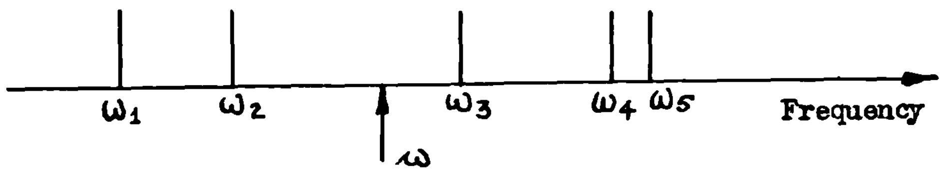


Fig. 2.8 Sketch Illustrating Mode Interaction

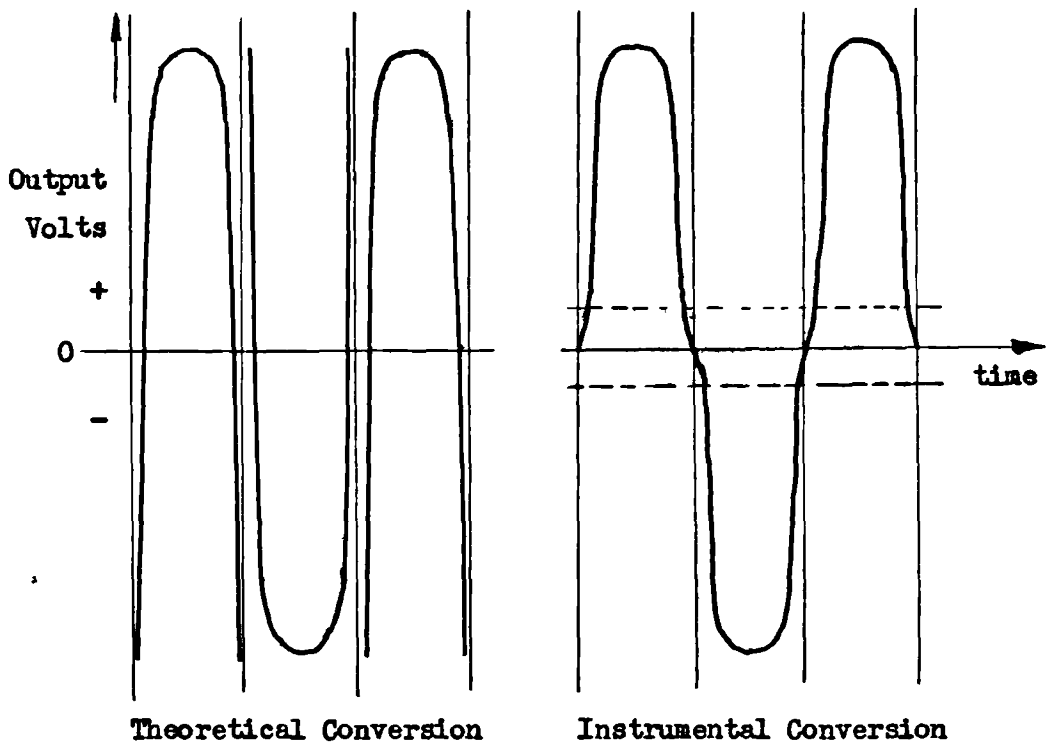


Fig. 2.9 Theoretical and Instrumental Conversion of a Sinusoidal Waveform. In the Instrumental Conversion, the law breaks down in the voltage range indicated by shading.

$$v_r = \frac{U}{Z} \cdot e^{i(\omega t - \phi_r)} \qquad \ldots\ldots\ldots (2.10)$$

where v_r = particle velocity due to rth mode

U = pressure strength of source

$$Z_r^2 = \frac{1}{\psi_r^2}\left\{R_r^2 + \omega_r^2 m_r^2 \left(1 - \frac{\omega_r^2}{\omega^2}\right)^2\right\} \qquad \ldots\ldots (2.11)$$

$$= \frac{1}{\psi_r^2}\left\{R_r^2 + m_r^2 \left(\frac{\omega^2 - \omega_r^2}{\omega}\right)^2\right\} \qquad \ldots\ldots (2.12)$$

and $\tan\phi_r = \frac{m_r}{R_r}\left(\frac{\omega^2 - \omega_r^2}{\omega}\right)^2$

where $\omega_r \cdot \frac{m_r}{R_r} = Q_r$, the Q factor for the mode.

Now, Bandwidth $= \frac{\omega_r}{Q_r} = \frac{\omega_r}{\omega_r} \cdot \frac{R_r}{m_r} = \frac{R_r}{m_r}$ = B, say

$$\therefore\ Z^2 = \frac{R_r^2}{\psi_r^2}\left\{1 + \frac{1}{B^2}\left(\frac{\omega^2 - \omega_r^2}{\omega}\right)^2\right\} \qquad \ldots\ldots (2.13)$$

$$\tan\phi_r = B^2 \left(\frac{\omega^2 - \omega_r^2}{\omega}\right)^2$$

Clearly, the excitation decreases as $(\omega^2 - \omega_r^2)$ increases, particularly if B is small, and in general the excitation decreases rapidly from the nearest mode to the more distant ones, and one would not expect the decay curve to be significantly influenced by individual modes other than the nearest two or three to the exciting tones.

We may write the real parts of the pressure components of the modes in the simplest form as

$$P_1 \cos \omega_1 . te^{-kR_1 t} + P_2 \cos \omega_2 . te^{-kR_2 t} + \quad ....$$

at the moment of cut-off of the tone, because all modes will, neglecting phase-lags due to high resistive components, be either in phase with the forcing tone or in antiphase, i.e. P_r is either positive or negative, and sine components will not exist.

We will now examine the summation of these terms for increasing numbers of modes up to three, as they would appear on a logarithmic level recorder or cathode-ray display.

Single Mode

$$P = P_1 \cos \omega_1 . t \; e^{-kR_1 t}$$

This function is converted to its nominal logarithm by the action of the logarithmic amplifier: nominally because no analogue device can produce an output of $-\infty$ as required if the input signal is zero. With the semiconductor logarithmic amplifier used by the author for values less than approximately 0.03 per cent of the maximum signal, the output is zero and we therefore have a waveform as in the sketch of Fig. 2.9.

In practice, the form of the zero crossing is of no significance because the logarithmic conversion is followed by peak rectification, taking account only of the maximum value of the 'signal waveform'.* Hence, the output of the logarithmic

* The errors at the zero crossing, however, are of great importance for other applications and preclude the use of this device, for instance, as an analogue multiplier of two signals, either of which may even become zero.

amplifier may be written: $\quad$ const. $- k\, R_1.\, t$

This is clearly a straight line with a slope proportional to R_1.

Two Modes of Approximately Equal Decay Time

$$(P_1 \cos \omega_1 t + P_2 \cos \omega_2 t)\, e^{-kR_1 t}$$

$$= \left\{\frac{P_1 + P_2}{2} (\cos \omega_1 t + \cos \omega_2 t) + \frac{P_1 - P_2}{2} (\cos \omega_1 t - \cos \omega_2 t)\right\} e^{-kR_1 t}$$

$$= \left\{(P_1 + P_2) \cos \tfrac{1}{2}(\omega_1 + \omega_2)t \cos \tfrac{1}{2}(\omega_1 - \omega_2)t - (P_1 - P_2) \sin \tfrac{1}{2}(\omega_1 + \omega_2)t \right.$$
$$\left. \sin \tfrac{1}{2}(\omega_1 - \omega_2)t \right\} e^{-kR_1 t}$$

Since $\dfrac{P_1 + P_2}{2} \cos \tfrac{1}{2}(\omega_1 - \omega_2)t$ is always multiplied by

$\cos \tfrac{1}{2}(\omega_1 + \omega_2)t.e^{-kR_1 t}$ and $\dfrac{P_1 + P_2}{2} . \sin \tfrac{1}{2}(\omega_1 - \omega_2)t$ is

multiplied by $\sin \tfrac{1}{2}(\omega_1 + \omega_2)t$ and these short-period factors are

are in quadrature, peak rectification will yield

$$\log\, (P_1 + P_2) \cos \tfrac{1}{2}(\omega_1 - \omega_2)t - (P_1 - P_2) \sin \tfrac{1}{2}(\omega_1 - \omega_2)t \;\ldots\; (2.14)$$

This will give maxima and minima of $\log\,(P_1 + P_2)$ and $\log\,(P_1 - P_2)$ respectively.

Clearly, this forms a sinusoid superimposed on a straight downward-sloping line, the slope being proportional to the band-width of the mode, i.e. inversely proportional to the reverberation time.

The amplitude of the sinusoid will be greatest if $P_1 = P_2$. In this condition, the signal will fall to the lower limit set by the law of the logarithmic convertor or by electrical noise in the circuit or acoustic noise in the enclosure. Exact equality will not however be maintained for long, in view of slight differences of decay rate, but dips in the decay curve amounting to 40 - 50 dB are sometimes observed, i.e. $(P_1 - P_2)/(P_1 + P_2) < 0.01$. Noise or other modes prevent this figure falling any lower.

On the pulsed glide display, the maximum to minimum ratio is usually small (near 1) close to each of the modal frequencies and reaches a maximum approximately halfway between the two modes.

Fig. 2.10 shows the curves calculated from Equations (2.10) to (2.14) for a typical case.

Two Modes of Unequal Decay Rate

This case may be treated by reference to the equal decay rate case. If during the course of the decay, the relative values of P_1 and P_2 change, the depth of the sinusoid will change. If the mode with the lower initial excitation decays the more slowly, the depth of modulation of the trace will increase during the decay and conversely if the less-excited mode decays more rapidly the depth of the modulation decreases.

Fig. 2.11 shows a set of traces calculated on the assumption that the lower-frequency mode has a decay rate twice that of the higher mode.

Three Modes

Three modes excited simultaneously with comparable initial amplitudes give far more varied and complicated decays than do two modes. The analytical treatment is also very much more difficult because the number of variables is greatly increased. Neverthe-

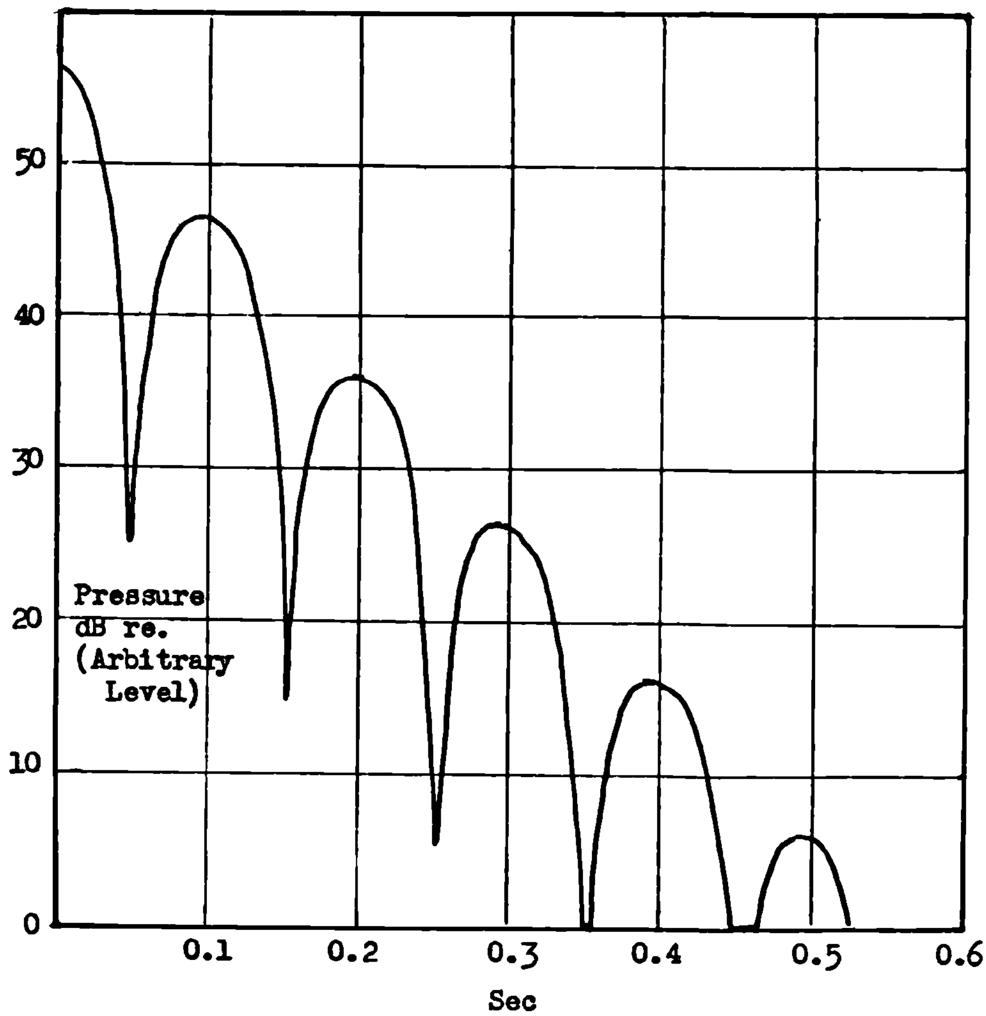


Fig. 2.10 Logarithmic Display of two Interacting Modes

$P_1 = 2.22$ $\qquad$ $P_2 = 0.222$

R.T. = 0.6 sec

$\dfrac{(\omega_1 - \omega_2)}{2\pi} = 10$ c/s

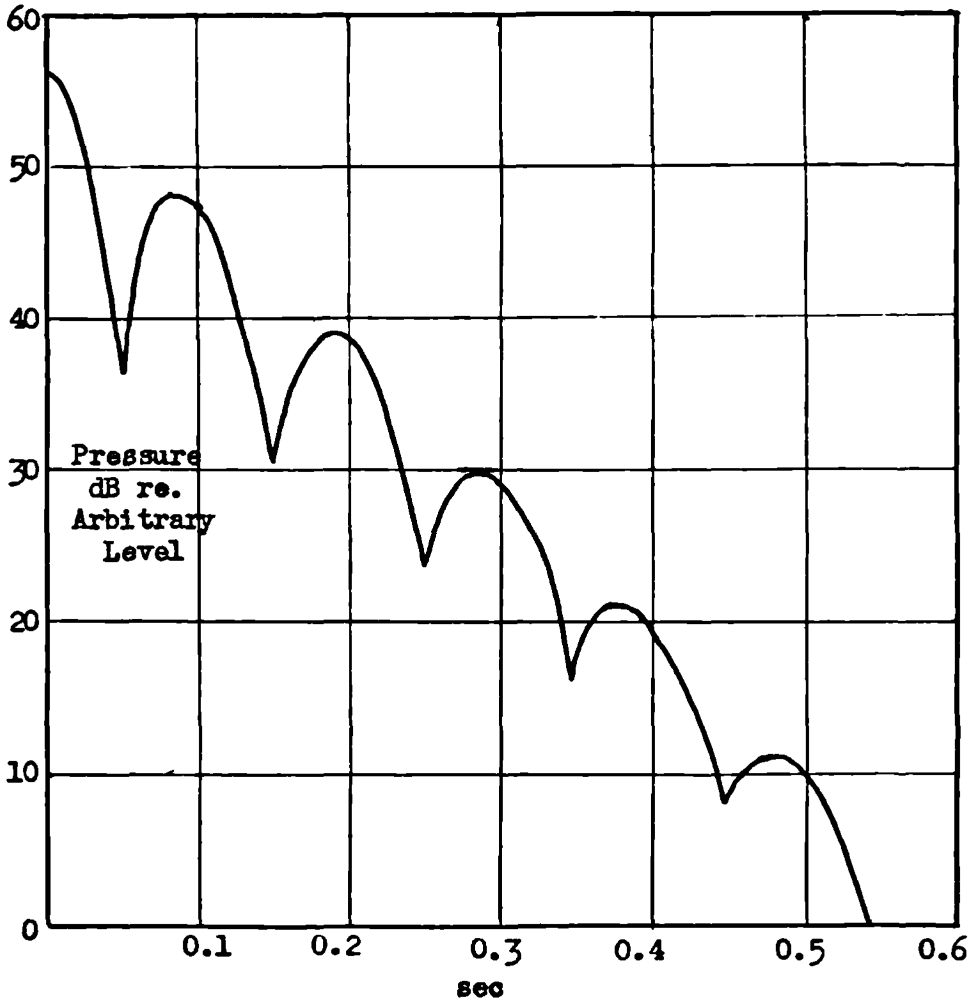


Fig. 2.11 Logarithmic Display of two Interacting Modes (unequal Decay Rate)

$P_1 = 2.22$ $P_2 = 0.222$ at $t = 0$
0.111 at $t = 0.5$ sec

$T = 0.55$ sec

less, it will be worthwhile to examine some special cases since it may in this way be possible to recognise three-mode characteristics in traces. To avoid complications, we assume that all relevant modes have the same decay time, so that the trace representing the logarithm of the pressure is represented by the superimposition of an oscillating function on a sloping straight line.

In a typical talks studio such as one has in mind for applications of this work, the reverberation time will be less than 0.6 sec. The pulsed-glide displays or the conventional level-recorder display will therefore, having a maximum range of 50 dB, show a portion of the decay curve up to 0.5 sec. in duration. It is of value before starting to calculate numerical cases to decide the upper and lower limits of interest. The lowest rate of fluctuation is that for which only, say, three complete fluctuations appear on the trace. Below this number it will be impossible without a very detailed study of the trace to distinguish a simple two-mode decay from one perturbed by the addition of a third mode. This places the lower limit of frequency separation at 6 c/s. The upper limit is placed by the point at which the modes are separated by such an interval that interaction between them is negligible.

If we pursue the same treatment as for two modes we represent the total pressure of the three modes by

$$P_t = P_1 \sin \omega_1 t + P_2 \sin \omega_2 t + P_3 \sin \omega_3 t$$

$$= K_1 (\sin \omega_1 t + \sin \omega_2 t) + K_2(\sin \omega_2 t + \sin \omega_3 t) + K_3(\sin \omega_3 t + \omega_1 t)$$

where $K_1 = \frac{1}{2}(P_1 + P_2 - P_3)$ etc.

Each of these terms is similar to the whole pressure due to two modes giving a product of sum and difference terms. After rectification and smoothing the sum terms disappear, leaving the difference terms which are in all practical cases of very low frequency.

The approximate form of a decay may thus be found by working out the excitation amplitudes of the three modes from their bandwidth and their separation from the exciting f frequency. The difference frequencies are then tabulated and their amplitudes K_1, K_2, K_3, calculated from the expressions given above. The problem is then treated as a combination of the three difference vectors.

If this is done for particular cases, the following observations are made. The majority of cases give simple beating traces similar to those of two modes,but with variations of period and depth of beat as the trace proceeds. If there are two nearly equally excited modes near the exciting frequency and a third at some distance, there may be marked curvature to the trace, either concave or convex upwards. This probably explains numerous instances observed in the past of decays having rapidly increasing slopes, a feature which cannot be explained by reference to unequal decay rates for different modes, which always must produce upward curvatures. Fig. 2.12 shows a photograph of a set of such downward-curving decays.

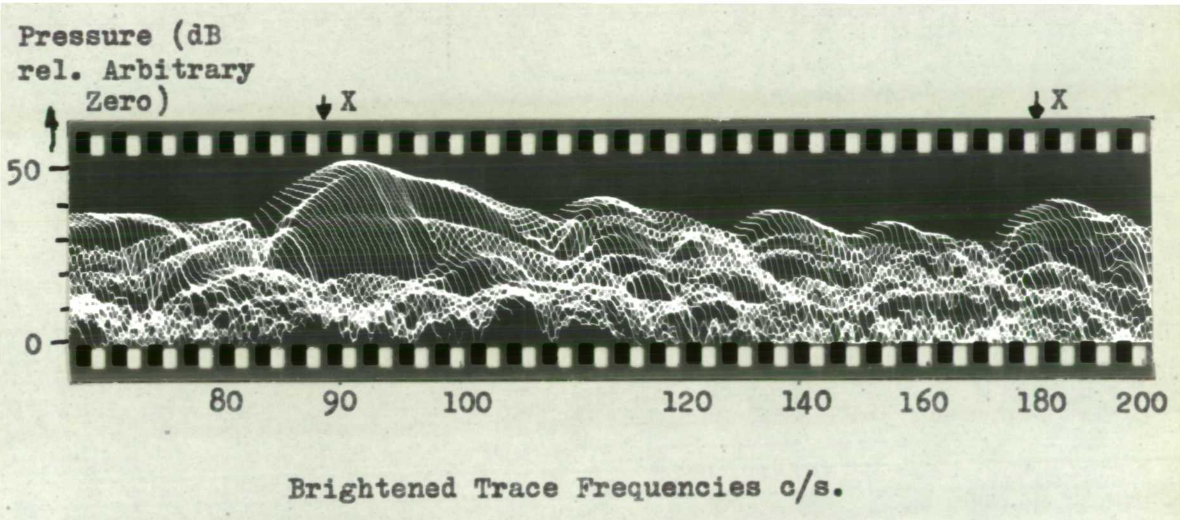


Fig. 2.12 Section of Pulsed Glide trace, showing downward-curving traces at regions marked X.

It would be an advantage to calculate a large number of examples of three-mode decays by this method, for classification as a means of recognising types just as two-mode decays can be interpreted by recognition of characteristics. A computer programme is now being tested by one of the author's colleagues to calculate and classify three-mode decay types.

CHAPTER 3

CONDITIONS FOR AUDIBILITY OF COLOURATIONS DUE TO ROOM MODES

Before proceeding to examine methods of displaying, assessing and reducing colourations, we will refer again briefly in this chapter to the conditions for audibility of a mode. Knowledge of these conditions has not progressed beyond the author's conclusion (Gilford 1959) and that it is the change of pitch during decay which is the most characteristic and disturbing feature of a mode which is heard as a colouration. Other properties have naturally been considered, such as a maximum of sound pressure at that frequency, or an increase of decay time for that mode compared with others. The pressure theory appears plausible because syllables spoken at the colouration frequency appear to be louder than at other frequencies, giving the effect of domination of that particular frequency; syllables at these frequencies may also appear to persist longer, lending support to the idea of a longer decay time.

Isolation, high excitation and long decay time are to a large extent interdependent and more than one may characterise a particular mode. It is therefore not easy to decide which of them is necessary and which merely contributive.

Since the publication of the paper referred to above, a considerable amount of evidence has accrued from the testing of new or re-treated studios. It is accepted practice to measure the reverberation time of the studio at half-octave intervals from 62 c/s to 8 kc/s using frequency modulated tone with a modulation depth of 7%. The low-frequency part of the audio-frequency spectrum is then explored using pure tone of which the frequency is slowly increased (at the rate of approximately one octave in two minutes). The sound decays are displayed on an oscilloscope, enabling the simple types of decay, that is those in which not more than two modes are effective, to be detected. The reverberation time of the room is measured at each modal frequency and compared with the curve obtained for the $\pm$ 7% bandwidth.

Even at those frequencies where colourations are subjectively observed, it is very unusual to find a significant difference between the reverberation time exactly at the modal frequency and the mean within the normal band of measurement. From what has already been written above it will be understood that the slopes of the single and two-mode types are very simple to measure since they consist respectively of unmodulated straight lines (on the logarithmic display) and successions of regular beats. The decays at non-modal frequencies are normally much less regular and more difficult to specify by a single slope.

Colouration thus does not appear to be a consequence of long decay time.

The remaining possibilities that colourations are due to high pressure-levels or pitch changes are not easy to disentangle. It is probable that both contribute to the sensation. For diagnosis and prediction, however, the more important of the two is the pitch change of sound at neighbouring frequencies, already mentioned, for the following reasons. Maxima of pressure occur at frequent intervals in the steady-state characteristic of the room. There seems no correlation between the heights of these maxima and the observation of colourations. This is particularly so if the microphone used for the measurement is as close to the loudspeaker radiating the tone (about 0.5 m) as the microphone is normally to the mouth of the speaker in a broadcast talk. In fact, the steady-state characteristic at this distance corresponds nearly to the combined characteristics of the loudspeaker and microphones, the room having very little influence. The pulsed-glide display at this distance shows features characteristic of the room, but at a very much lower level than the steady state. At this distance pitch changes can still be heard by the use of compression in the listening circuit (see Chapter 5 below). The sensation of colouration is therefore associated with changes in the signal which must be practically independent of level, which indicates that pitch change is the most important objective phenomenon.

Thus we may state the conditions necessary for colourations to be heard as follows.

(1) The mode must be axial and separated from its neighbours by 20 c/s or more. A cluster of such modes within an interval of about 5 c/s may be regarded for this purpose as a single mode.

(2) The axial modes of nearly rectangular rooms may be treated, as far as prediction is concerned, as has been shown in Chapter I, as rectangular rooms with dimensions equal to the mean height, length and width.

(3) The frequency must agree with a prominent frequency of the speaker's voice. Thus different speakers vary widely in their excitation of the colourations in a particular room.

(4) The neighbouring non-axial modes must not be too numerous. In a well-designed room this will restrict colourations to frequencies below 300 c/s. If however there are gross differences in the reflection coefficients of the three pairs of surfaces, this limit will be raised.

These conditions must be treated as still provisional. An attempt has been made during the course of this study to obtain conclusive verification but the difficulties of the subjective observations have unfortunately prevented this.

CHAPTER 4

METHODS OF DISPLAY OF ISOLATED MODES

4.1 General

It has been shown above that isolated modes are characterised by the absence of beats and by frequency shifts. Both these characteristics can be demonstrated by cathode-ray displays, one form of which was devised by the author and described by Gilford and Somerville (1951).

Pulses of tone of successively increased frequency are radiated into an enclosure and picked up by a microphone. After amplification the microphone output is applied to a logarithmic amplifier connected to the Y plates of an oscilloscope, the X-sweep of which is triggered at the end of each pulse. For a single room mode, the display will be a straight line, as shown in Chapter 3 above and the same will be substantially true of more complex conditions, though the line will be modulated by beats of random fluctuations. A series of simple linear decays is shown in Figure 4.1. Successive decays with increasing frequency are photographed side by side on a slowly moving 35 mm. film.

This process produces a composite display in which the nature of the changes of decay shape with frequency can be clearly seen. In particular, the following characteristic formations may easily be recognised:

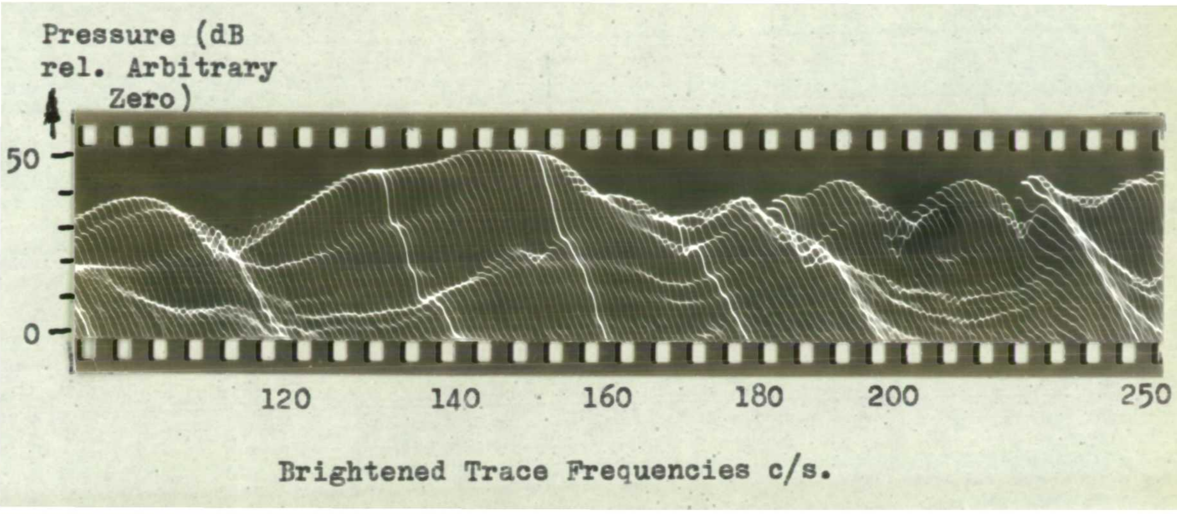


Fig. 4.1 Typical section of Pulsed-glide trace, showing various common formations.

(1) The passage through an isolated mode showing unfluctuating decays which appear as a region crossed by a series of parallel smooth straight lines. On each side of this frequency the decays develop beats of increasing depth.

(2) The existence of regions where the decays experience an abrupt change of slope in the later part of the decay.

These formations are caused more frequently by the presence of resonators of high Q-factor than by room modes with long decay times. They have been found to be due to Helmholtz resonators, wall or door structures, glass lampshades and metal objects.

(3) Regions of curved decays due to poor diffusion.

Figs. 4.2 (a), (b) and (c) show examples of these three types of display.

Fig. 4.2 (d) is an example of a cancellation of the reverberant by direct sound, due to the two components being substantially in antiphase. It will be seen that the initial level at the time of the cut-off of the tone pulse may be 15 dB below that reached subsequently during the reverberation. The resultant level at cut-off reaches a sharp minimum at 83 c/s, indicating that there is a sharp maximum of reverberation at that frequency.

Such a formation is therefore indicative of an isolated mode, even though it does not show the straight decays characteris-

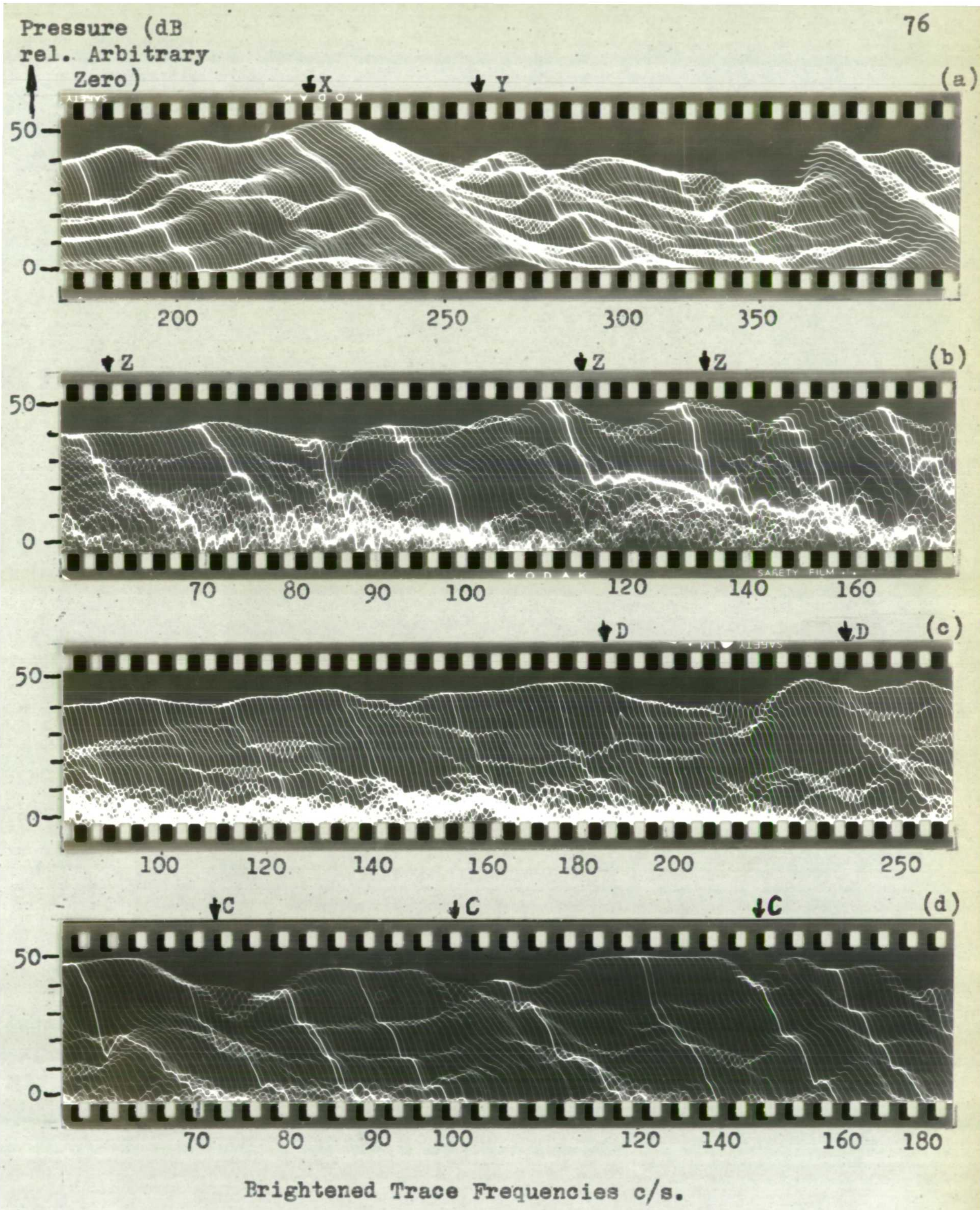


Fig. 4.2 Pulsed-glide traces.

(a) Showing linear decays typical of single mode at X,
and beats typical of two interacting modes at Y.
(b) Showing an abrupt change of slope at Z.
(c) Showing curved decays due to poor diffusion at D.
(d) Showing cancellation of reverberant by direct sound at C.

tic of a mode. This illustrates the fact that the pulsed-glide display may vary from point to point in a room, according to the positions of the loudspeaker and the microphone in relation to the boundaries of the room and to each other.

To overcome this ambiguity and to make the displays more meaningful, several devices have been tried. It is desirable to place the microphone to pick up reverberant sound of as many modes as possible. The obvious solution is to use a microphone with an omnidirectional characteristic in a corner of the room, choosing a corner which is simple in shape and free from skirtings, sound absorbing materials or apertures. The loudspeaker, similarly, should occupy a position from which it can excite as many modes as possible. A small loudspeaker can be placed in a corner and thus excite all modes for which its effective distance from each of the surfaces forming the corner is small compared with the wavelength. In practice, however, this limits the frequency bandwidth over which a particular loudspeaker may be used for the following reason.

To obtain a satisfactory pulsed glide, the level of the tone at cut-off must be at least 30 dB above the ambient noise in the studio. The effect of ambient noise is normally limited by octave-bandpass filters in the microphone circuits, but in any practical situation the noise-power in an octave bandwidth is much higher at low frequencies than at high.

In practice this requires a loudspeaker of large diameter, say 25 cm, and to maintain an adequate output at low frequencies, the loudspeaker will require a volume of 0.2 m^3 enclosed behind it and it is therefore not generally possible to place its radiating surface within a short enough distance of a corner to excite all modes at the higher frequencies. If we are solely concerned with colourations at low frequencies this will not matter, but in general one is concerned also with the possibility of rings at mid voice frequencies. An alternative position of the loudspeaker from which it excites all modes significantly (though not equally) is at a position one-third the distance along a diagonal ~~diameter~~ connecting opposite pairs of corners, e.g. (0,0,0) and (1,1,1). In such a position there can be no nodes of pressure for the lower orders of mode.

4.2 "Coherent" Glide Displays

An attempt was subsequently made to refine the pulsed-glide displays with the object of producing a display which was more characteristic of the room as a whole and less dependent upon the position of the source or the microphone.

The pulsed-glide display is dependent only on the amplitude of the sound pressure at the microphone and is unaffected by phase changes. In the modified form of the display, the microphone signal was modulated by the signal being fed to the source.

This gave as the amplitude of the display the product of the source input voltage, the microphone output voltage and the cosine of the phase angle between them.

If the microphone signal remains at the same mean frequency as the source, the phase angle will remain constant and the display will reflect only changes in the microphone signal. If, however, there is a difference between the mean frequency of the microphone signal and the source during the decay, the decay will be modulated by a beat with a frequency equal to the difference between the source and received signal frequencies.

To display these phenomena, the tone and amplitude axes as normally arranged on the pulsed glide displays were interchanged, the cathode-ray trace traversing the film from edge to edge with advance of time and being deflected along the length of the film by modulated signal.

Details of the technique and results are given in a B.B.C. Monograph by Gilford and Greenway (1956). The conclusions from the monograph are that

(1) The display permitted the separation of modes as close together as 2 c/s, while the normal pulsed-glide display was unable to achieve this.

(2) Single isolated modes gave characteristic patterns, though they were no easier to recognise than on the conventional display.

4.3 <u>The Coherent Counter</u>

In spite of these conclusions, this method remained the only available method of displaying differences in frequency between the originating source and the decaying sound-field, which the author felt was the essential feature of colouration.

A modification was therefore made to the coherent-glide apparatus to enable the beats, or reversals of phase, to be counted during the decay. The output of the modulator was amplified, the peaks clipped to reduce the oscillations to square waves which were then differentiated, giving alternative positive and negative pulses which were then used to operate a mechanical counter.

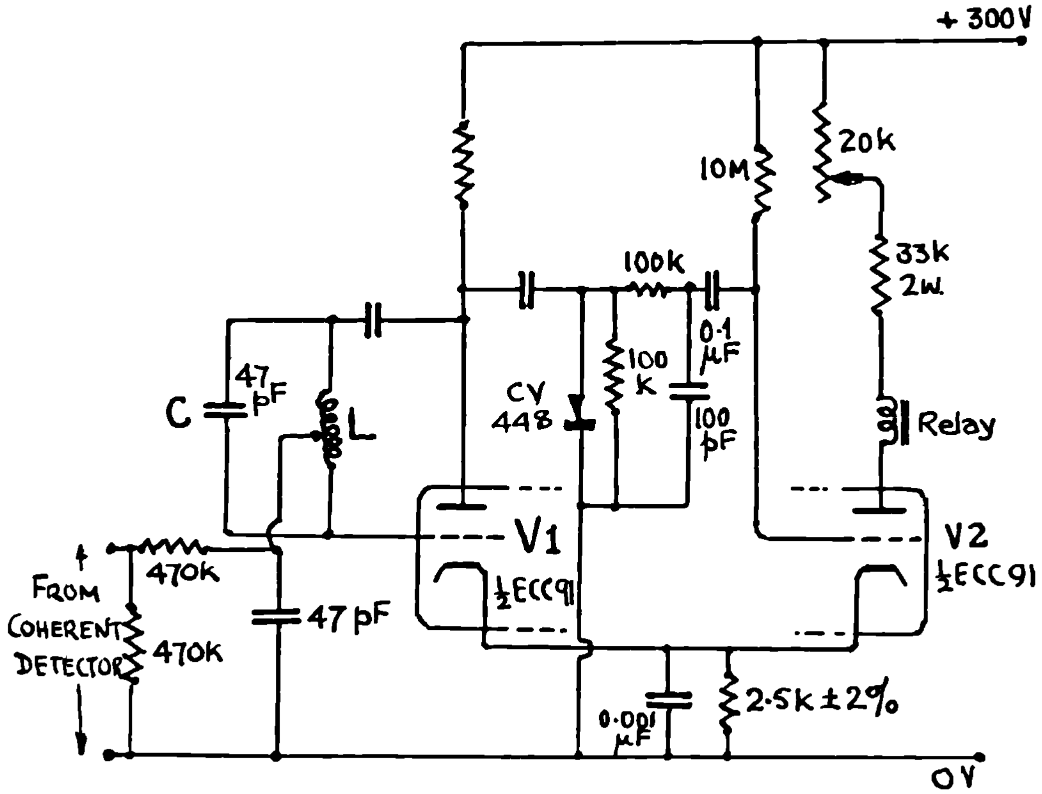
A second counter was operated by the pulses constituting the source input, so that the ratio of the readings of the two counters over a particular frequency interval shows the average number of zero-crossings of the coherent-detector output per test pulse.

The primary object of this device was to obtain a quantitative measure of the extent to which a mode causes a pitch change in test signals and hence the extent to which it would be appreciated as a colouration of the programme.

The second object was to obtain a statistical figure of merit for any particular enclosure. The mean ratio of the two counter readings over the range of frequency from the lowest significant voice frequency to the limit of isolated mode behaviour (say 50 c/s to 300 c/s) would be expected to serve as such a criterion.

Fig. 4.3 shows the circuit used for the zero crossing counter.

Experiments were made by Miss S. Edwards, using this device in an enclosure of volume 1500 ft^3. The acoustic treatment could be varied to give reverberation times between 0.3 sec and 1.2 sec by means of demountable absorbing panels. To make a



V1 in conjunction with the tuned circuit C,L is a Hartley oscillator but is biased off by the potential maintained on the cathode of V2 whose grid is connected through a 10 M resistor to the H.T. line. If the input on the grid of V1 rises to 1 volt, V1 starts to oscillate. The output is rectified by the diode CV 448 and differentiated by the 100 k resistor and 100 pF capacity, supplying a negative pulse to the grid of V2 which cuts off and operates the counter relay in its anode, and further reduces the bias of V1. The 0.1μF blocking condenser prevents continuation of the process and V2 returns to steady state to await the next pulse.

Fig. 4.3 Circuit of Reversal Counter
(devised by M.W. Greenway)

permanent record of the counter signals, they were also fed to a pen recorder marking on a slowly moving waxed paper. A separate push button connected to the audio frequency oscillator enabled frequency markers also to be manually impressed on the waxed paper. The oscillator was glided through the frequency range of interest in the same manner as the normal pulsed or coherent glides, giving equal lengths of paper chart for each successive octave of frequency change.

Fig. 4.4 is a reproduction of typical traces obtained by the method. The figure shows the traces from four different microphones in the room; it will be seen that at a few frequencies, approximately 90 c/s, 140 c/s and 175 c/s there are groups of reversals common to all positions, and these are the frequencies at which colourations had already been noted in speech tests. In attempting to put the test on an absolute quantitative basis as suggested above, however, less success was achieved. The obvious disparity between the behaviour of several microphone positions was the first objection since it was clear that the results from many positions would require to be averaged in order to obtain a significant result. This would make the test very time-consuming and means of combining the outputs of several microphones were devised.

Two methods were tried, both combining three microphones at different positions in the room. The simple notion of adding the three outputs by means of a star network or other resistive

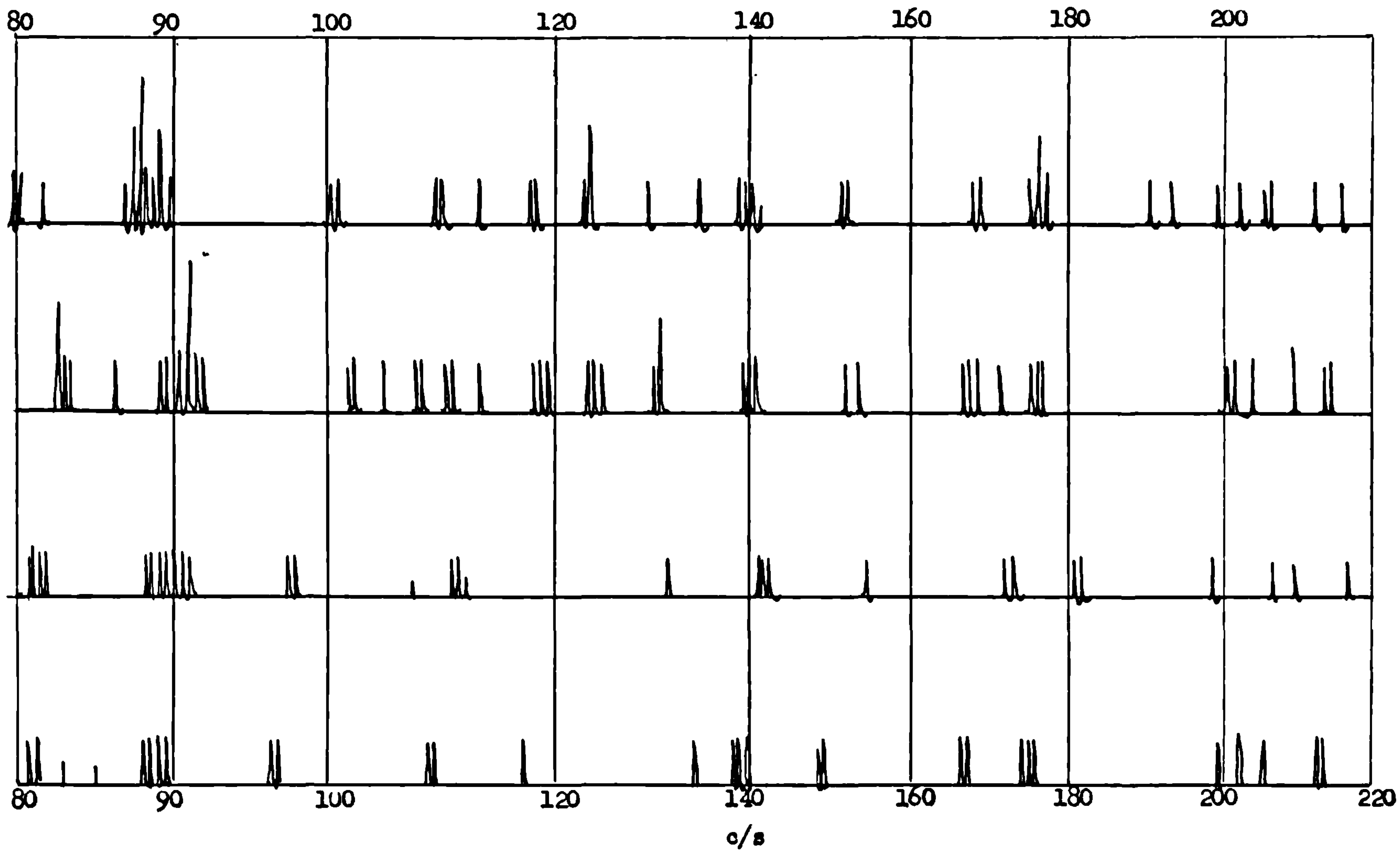


Fig. 4.4 Coherent Counter Traces from Four Arbitrary Microphone Positions in a Small Talks Studio.

circuit would not be valid because the signal from the three would be phase-coherent and the sum would therefore, with varying frequency, shows a series of maxima and minima which were dependent only on the relative positions of the microphones in the room. For instance, if one microphone were at a maximum of the standing-wave pattern at a particular frequency and the other two were at positions where the sound-field was in anti-phase with that at the first position, total cancellation of the signals could result, even if the standing-wave system is an important one associated with strong colouration phenomena. (Indeed, it appears from an examination of the mathematics of vector addition that virtually complete cancellation is more probable where only equally or oppositely phased components are of significance than if relatively strong components at other angles, such as those due to the direct exciting signal and its early reflections, are present.)

To avoid these unwanted effects it would be necessary, therefore, to add the rectified envelopes of the three signals and operate on the sum thus produced. This would, however, remove the frequency information from the microphone signals and with it the basis of the coherent display.

One possibility remained - the three signals could be individually full-wave rectified, giving three-wave trains in the same phase but of twice the original frequency. The sum of these could then be compared with a full-wave rectified original tone radiated into the room. There would be large fluctuations in the sum as the frequency was varied, which would be a function not only of the excitation of the mode but of the positions of the microphones in the standing-wave patterns at the modal frequencies, but reversals and cancellations would not be present. Nevertheless, owing to these random fluctuations it was doubtful whether the method would be really satisfactory.

For these reasons, it was decided to test the validity of the results of the coherent counter using corner positions for a single microphone, before assembling circuits for the combination of several.

Careful tests with a single microphone gave very variable results which appeared to be very sensitive to loudspeaker position and output. These uncontrollable variations were large enough to obscure the difference between counts in the same room in its initial state and after removal of a sufficient quantity of absorbing material to alter the acoustics noticeably.

It was therefore clear that attempts to compare one room and another would be bound to fail. By doing repeated tests in the same room in two different acoustic conditions it might be possible to assign a figure of merit and a variance to each condition, but if the whole experiment were moved to another room it would be impossible to relate the loudspeaker positions to that in the first room. The relationship would have no meaning since the standing-wave fields in the two rooms would be completely different.

This method was therefore finally abandoned, and it was decided to explore methods more closely related to actual speech conditions, with particular attention to spectrum displays as described in the next chapter.

CHAPTER 5

METHODS OF ASSESSING COLOURATIONS

5.1 General

In the last section, attempts to establish probable frequencies of colourations were described in some detail, with reference to earlier work by the author, some of which has not hitherto been published. This section deals with methods depending on subjective tests or tests using speech as the test signal. Work carried out concurrently with the objective methods of the section above will be briefly mentioned and recent work using narrow band spectral analysis, not yet brought to completion, will be described in full.

5.2 Purely Subjective Methods

To a certain limited extent, the frequency and severity of a colouration may be assessed by straightforward listening to speech over a microphone and loudspeaker. This is, indeed, the test by which final judgment is made on the merit of a speech studio. Some refinement of this method is, however, needed for the following reasons:

(1) The severity of the colouration varies greatly with the position of the microphone in the room, with the directional characteristics of the microphone and the spectrum of the speaker's voice.

(2) A quantitative scale of severity is required in addition

to a statement of the frequency in order that the effect of any remedial measures taken to effect an improvement may be confidently assessed.

(3) Remedial measures often reduce the main colouration, allowing a less important one at another frequency to be audible. It would be valuable to know of the existence of both in the first place.

As an aid to direct subjective assessment, use may be made of a narrow-band selective amplifier. Preliminary experiments showed that colourations could be made more easily by adding to the reproduced sound a proportion of the original microphone output which had been modified through an amplifier containing a frequency-selective filter which passed only a narrow band of frequencies, the centre of which could be continuously varied within wide limits. The first tests were made with a proprietary amplifier and as a result of the experience gained, a more convenient instrument was designed by M. W. Greenway (1953). Fig. 5.1 represents a block schematic diagram of the amplifier and its use for the detection of colourations. The amplified microphone signal is applied to the inputs of an amplifier A which has a negative feedback loop incorporating a Wien Bridge network which has a sharp maximum of impedance at a particular frequency. At this frequency, which may be tuned over a wide range by altering the values of two ganged variable resistors, the negative feedback is reduced and the gain of the stage is thereby increased.

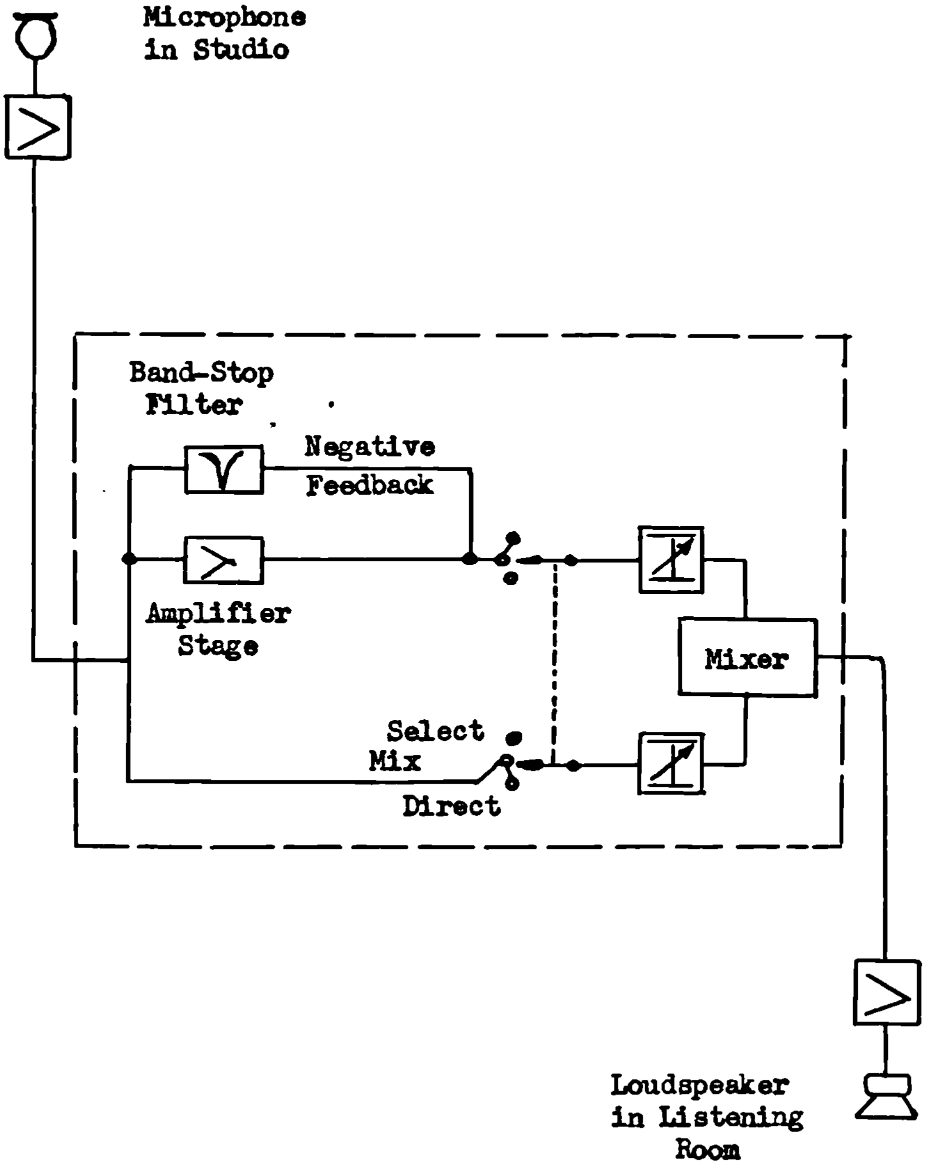


Fig. 5.1 Selective Amplifier for the Detection of Colourations in Speech.

The output of this stage goes to a second stage from which an independent negative feedback is taken to the input of the selective amplifier, reducing the gain of the two stages together at the selected frequency to unity. These two stages, therefore, form an amplifier which gives an output only close to the tuned frequency with a bandwidth of ten per cent of that frequency. The output is connected to a switch to which is also taken a parallel connection straight from the input. This switch connects one or both of the circuits to separate attenuators with a range of 40 dB, the outputs of which are mixed by being connected to the two grids of a double triode valve with the anodes in parallel. At the tuned frequency there is no phase difference between the 'direct' and 'select' signals.

The mixed output is amplified by a power amplifier and fed to a loudspeaker in an acoustically treated listening room.

The method of use is to connect the selective amplifier and loudspeaker to a preamplifier connected to a microphone in the room under investigation, and to listen to a speaker in the studio reading continuously to the microphone.

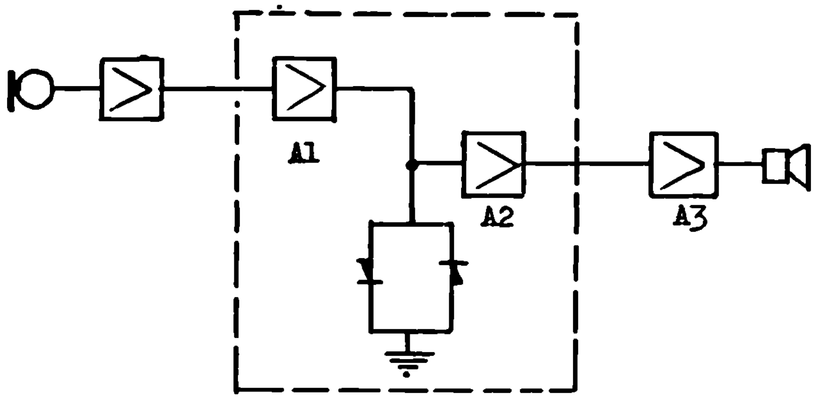
The attenuator in the 'selective' circuit is turned down some 10 dB below the preferred listening level on the direct channel and the amplifier is then tuned slowly and continuously from 50 c/s to 500 c/s. If no position in the frequency scale produces a prominent colouration on the speech, the 'select' attenuator is adjusted to increase the level by 2 dB and the listening repeated. This process is continued until a frequency is found at which there is a noticeable colouration due to the reinforcement of an existing colouration by the selective amplifier. By further increasing the level in the selective circuit other colourations can be found which would not necessarily be noticeable in the presence of the primary colourations.

This method has been found very useful over a long period, though it has all the disadvantages of requiring the co-operation of a speaker as well as listeners, and very close concentration by the listeners for a long period. It also requires some skill and experience and cannot therefore be regarded as a fully satisfactory method for general use.

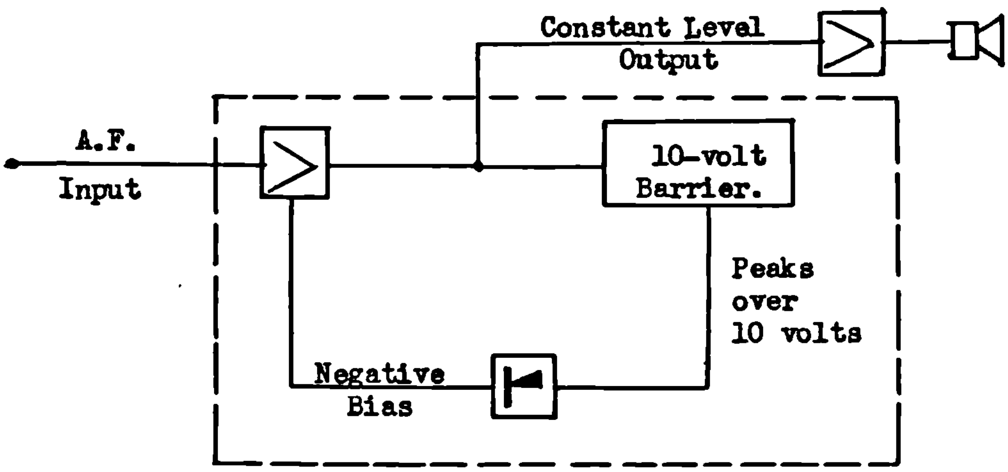
5.3 Methods Using Amplitude-Compression

A method in which listening was assisted in another way has been used with some success, and the principle will now be described. In rooms of the size of those with which we are here concerned, the reverberation time is so short that it is difficult to hear changes in the quality or pitch taking place during the period of decay, particularly if they occur during the later stages when the sound is approaching the level of inaudibility.

Any "compression" of the sound signal, i.e. amplification which increases as the sound level falls, naturally assists hearing of changes throughout the decay until it reaches the background noise level. A certain improvement can be gained by the use of a non-linear amplifier such as that shown in Fig. 5.2 (a). The amplified microphone signal is fed to a high-impedance



(a)



(b)

Fig. 5.2 Devices for Making Pitch-Changes Audible in Sound Decays by Compression

(a) Simple Compressor with Logarithmic Law
(b) Constant-Level Listening Device

amplifier A_1, the output of which is taken to earth through a pair of silicon or germanium diodes DD which are in parallel to each other, but opposite in direction. The high-output impedance of A_1 ensures that its output current is substantially unaffected by the comparatively low forward impedance of the diodes. DD are matched diodes of a selected type (e.g. Mullard OA1) of which the voltage established across the diode by a current in its forward direction is approximately proportional to the logarithm of the current. This relationship holds for a range of currents as great as 1000 to 1, i.e. 60 dB. The potential difference across the diodes will change in sign with the periodic reversals of the input signal each diode in turn conducting whilst the other is non-conducting and thus the potential is the logarithm of the modulus of the microphone signal with the same instantaneous sense throughout. This potential is amplified by a conventional amplifier circuit and fed to a loudspeaker amplifier A_3 for listening tests.

A more powerful method was devised, making use of an existing logarithmic amplifier (Mayo, Beadle et al, 1951) developed in the laboratory in connection with the pulsed glide displays described elsewhere in this paper. The principle of operation of this amplifier illustrated by Fig. 5.2 (b) is that the bias on a valve with a highly non-linear grid characteristic is controlled by the feedback from a later stage in the amplifier. This stage incorporates a diode maintained at a potential of 10 volts and any peaks of voltage exceeding 10 volts which reach the diode are able to pass to the rectifying and smoothing circuits supplying the first-stage bias.

The anode of the valve feeding the 10 volt barrier diode therefore maintains a virtually constant peak signal level and by connecting a loudspeaker circuit to this point it is possible to hear the decay of a sound within a room at constant level from the start until it is obscured by background noise at the end of the decay. Pulses of tone about 0.6 sec. long and of gradually increasing frequency are radiated into the studio under test at intervals of approximately 2 sec. The decay of the sound is heard as a tone of constant level but changing timbre until it is suddenly masked by random noise, the amplifier operating at full gain because of the decay of the biasing signal.

In the neighbourhood of prominent isolated modes, the tone is heard to change in pitch during the course of the decay; as the frequency of the exciting tone rises towards the modal frequency it glides downwards during the course of the decay. In this way it is possible to identify prominent isolated modes with very little difficulty and these are often found to be associated with colourations identified by the more laborious and skilled methods of direct listening.

5.4 Spectrometer Method of Assessing Colourations

5.4.1 Theoretical Considerations

The conclusion, reached as a result of work described above on the nature of the colouration phenomenon, is that its essential feature is a transference of energy during the reverberation process to particular frequency bands from neighbouring ones. The bandwidths concerned are of the order of 5 to 10 c/s, this width being appropriate to a reverberation time of a small speech studio, and its approximate value confirmed by the spacing between prominent modes found experimentally to be necessary for a colouration to be heard.

The possibility therefore presents itself that if the spectrum of any syllable of speech were analysed into a series of bands of the order of only 10 c/s in width, colouration frequencies would appear as high bands in the spectrum. These high regions might be directly observable in a display of the spectrum or it might be necessary to compare the spectrum of speech recorded in a non-reverberant room with that of a recording of the same speech played into the room under investigation.

Audio-frequency spectrographs are not new in principle. Messrs. Standard Telephones and Cables made a few examples before 1940 but the bandwidth of each filter was one third of an octave, amounting to approximately 25 c/s at 100 c/s and increasing in width at higher frequencies. Clearly this type would be unsuitable. Potter and Steinberg (1949) have also constructed a speech spectrograph in which the spectrogram is continuously displayed against time. A two-dimensional diagram is produced as a cathode ray oscillograph, the horizontal axis representing the passage of time, the vertical axis being a frequency scale and spectral density being represented by beam brightness. By photographic methods permanent records of the "visible" speech patterns of syllables and words can be made.

Here, again, the bandwidths of the filters were far in excess of the 10 c/s required for the present work. A more recent spectrograph constructed at the Post Office Engineering Section (Gill 1961) uses filters of 50 c/s bandwidth.

The analysis of speech with narrow bandwidths presents a fundamental difficulty because each individual syllable will give different spectra it is therefore necessary that the apparatus should perform a complete analysis during the period of a single syllable, say in 0.15 sec. (Bolt and Macdonald 1949). Colourations of frequencies less than 80 c/s have seldom been recorded since there is little energy in the speech spectra, even of male voices, below that frequency. Frequencies above 300 c/s are normally free from singularities though very obvious colourations even as high as 1000 c/s may occasionally occur. We therefore need to cover 220 c/s in 10 c/s bands, and to provide 22 filters for the analysis. The response time of each filter must therefore be 0.15/22 sec. i.e., 1/147 sec.

It would be conceivably possible to design a filter similar to the selective amplifier described in section 5.2 above and to scan the frequency range repeatedly by means of ganged continuously variable resistors. However, if this were done, each band of 10 c/s would take only 1/147 sec. to scan, during which time only ½ cycle of the lowest frequency or 2 of the highest frequency would elapse. The sampling of such small numbers of cycles imposes a very large uncertainty (or large effective bandwidth) on the frequency, as will be shown by the following calculations.

We require to know the width of the power spectrum of a very short pulse of tone containing only one or two complete cycles. To do this we evaluate the Fourier transform of the time function consisting of n complete cycles of a sinusoid represented by $P(t) = P_o \sin\omega_o t$ and bounded by fixed starting and stopping times t_1 and t_2.

This transform is given by

$$F(\omega) = \int_{t_1}^{t_2} e^{-i\omega t} P(t)\, dt$$

For the sake of symmetry, we can, witbout loss of generality, put $t_2 = -t_1 = \tau/2$, so that, substituting $\sin\omega t = \frac{e^{i\omega t} - e^{-i\omega t}}{2i}$ we have

$$F(\omega) = \frac{P_o}{2i} \int_{-\tau/2}^{+\tau/2} e^{-i\omega t} (e^{i\omega_o t} - e^{-i\omega_o t})\, dt$$

$$= -iP_o \left\{ \frac{\sin(\omega-\omega_o)\tau/2}{\omega-\omega_o} - \frac{\sin(\omega+\omega_o)\tau/2}{\omega+\omega_o} \right\}$$

Case 1

Let us consider first the case where the wave-train starts and finishes at times where the function $P(t) = 0$, i.e. at times $\pm \frac{n}{\omega_o}$.

$$\text{Then } F(\omega) = -P_o i \left\{ \frac{\sin(\frac{\omega}{\omega_o} - 1)n\pi}{\omega-\omega_o} \quad \frac{\sin(\frac{\omega}{\omega_o} + 1)n\pi}{\omega+\omega_o} \right\}$$

which may be reduced to

$$F(\omega) = \frac{-2i\omega P_0}{\omega^2 - \omega_0^2} \cdot \sin \frac{\omega}{\omega_0}\pi n$$

Since when $\omega = \omega_0$ this result is indeterminate, we investigate the limiting value of the expression when $\omega = \omega_0 + \Delta$ and $\Delta \to 0$

$$\mathop{\mathrm{Lt}}_{\Delta\to 0}^{} F(\omega_0 + \Delta) = \mathop{\mathrm{Lt}}_{\Delta\to 0} \frac{-2i(\omega_0 + \Delta)P_0}{2\omega_0\Delta + \Delta^2} \sin n\pi\frac{\omega_0 + \Delta}{\omega_0}$$

$$= \mathop{\mathrm{Lt}}_{\Delta\to 0} \underline{-P_0 i} \cdot \sin \frac{n\pi\Delta}{\omega_0}$$

Now if $f_1(\Delta), f_2(\Delta) \to 0$ as $\Delta \to 0$,

$$\mathop{\mathrm{Lt}}_{\Delta\to 0} \frac{f_1(\Delta)}{f_2(\Delta)} = \mathop{\mathrm{Lt}}_{\Delta\to 0} \frac{f_1'(\Delta)}{f_2'(\Delta)}$$

then $\displaystyle \mathop{\mathrm{Lt}}_{\Delta\to 0} \frac{-P_0 i}{\Delta} \cdot \sin\frac{\pi n\Delta}{\omega_0} = -P_0 i \mathop{\mathrm{Lt}}_{\Delta\to 0} \frac{n\pi}{\omega_0}\cos\frac{\pi n\Delta}{\omega_0}$

$$= \pm \frac{P_0 i n\pi}{\omega_0}$$

Hence $\displaystyle \left|\frac{F(\omega)}{F(\omega_0)}\right| = \frac{2\omega_0^2}{\omega^2 - \omega_0^2} \cdot \frac{\sin\frac{\omega}{\omega_0}\pi n}{\frac{\omega}{\omega_0}\cdot\pi n}$

<u>Case 2</u>

Now consider a wave train started and stopped at its maximum positive or negative value, such as $\tau/2 = (n + \frac{1}{2})\pi/\omega_0$.

Working similar to that for Case 1 gives the expression

$$\frac{F(\omega)}{F(\omega_0)} = \frac{2\omega_0^2}{\omega^2 - \omega_0^2} \cdot \frac{\sin \frac{\omega}{\omega_0}\pi n}{\pi n}$$ which differs

from the first case only in the substitution of ω_o^2 for $\omega\omega_o$ in the first factor.

This difference represents in relation to the first case a halving of $F(\omega)$ for every octave of frequency and hence a flatter spectrum. Any start and finish times intermediate to these cases will be represented by linear combinations of the two and will therefore give spectra intermediate between them.

Hence the narrowest possible spectrum for a particular value of n is given by Case 1. If n=2, representing the example quoted above for a filter with centre frequency 300 c/s scanned in approximately 1/150 sec., the spectrum will be as shown in Fig. 5.3. The bandwidth is clearly far greater than 10 c/s, and the method of varying the tuning of a single filter will therefore not be admissible.

An alternative, involving a great many more components and greater difficulty in the design of the filter circuits is to provide a series of independent filters, each having a bandwidth of 10 c/s and a different central frequency, and all simultaneously excited by the programme. A switch then samples the signal after transmission through each of these filters in sequence, thus building up the spectrum once during each rotation of the switch.

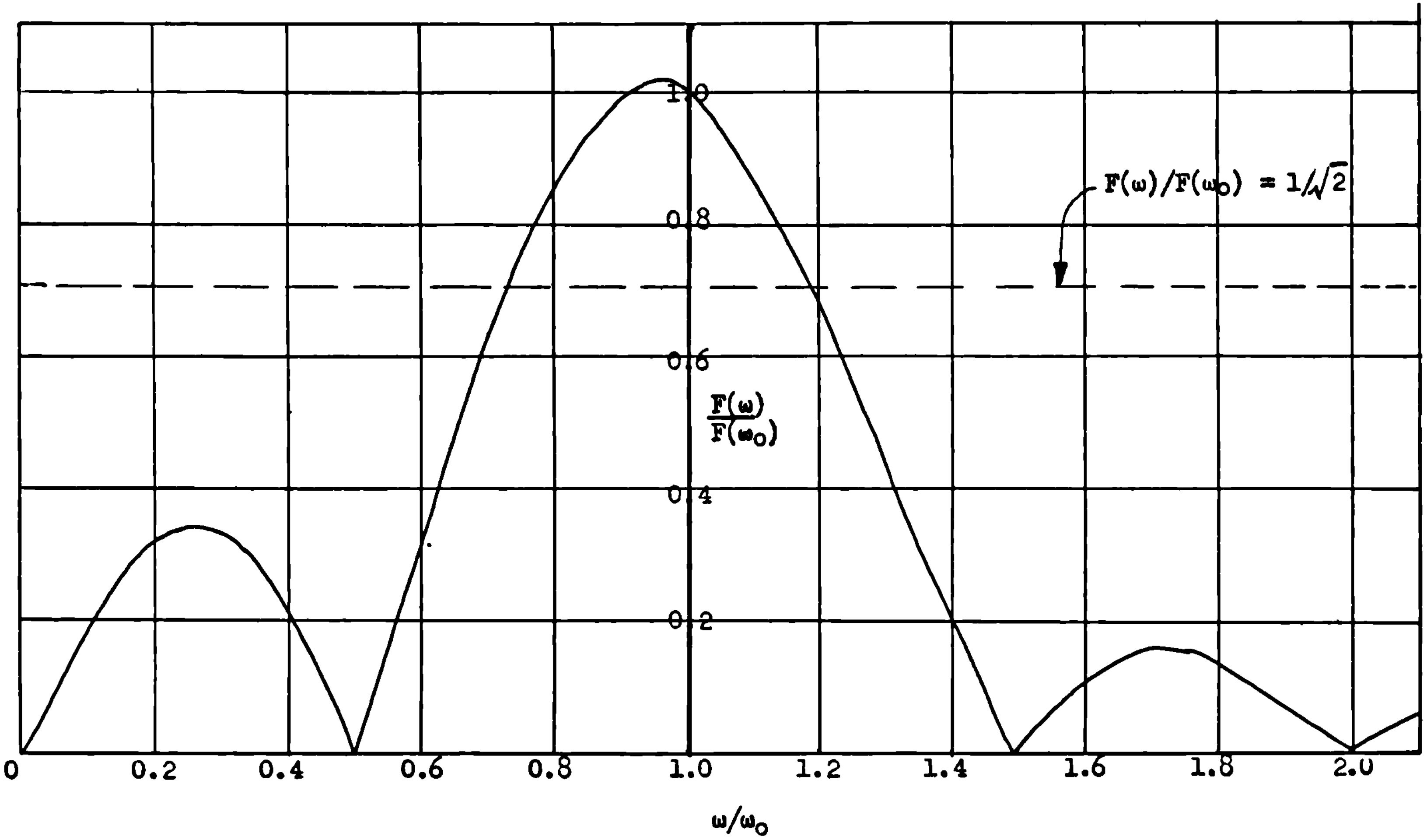


Fig. 5.3 Spectrum of Short train of Sinusoidal Signal. Two complete cycles.

One rotation per syllable would for this purpose be enough to give a close approximation to a continuous display of the spectrum on a cathode-ray oscillograph.

We therefore now investigate the length of time, or number of cycles, required to produce a spectrum in bands of 10 c/s over the range of frequencies from 80 to 300 c/s.

The bandwidth of a filter may be defined as the frequency interval between the two points at which its response has dropped to $1/\sqrt{2}$ of its peak response, this being the half-power point for the spectrum.

Here, $$\frac{F(\omega)}{F(\omega_0)} = 0.707$$

At the extremes, 80 c/s and 300 c/s respectively, the half-bandwidth, 5 c/s represents values of $\frac{\omega}{\omega_0}$ given by (1 ± 0.062) and (1 ± 0.0167) respectively.

Now, we have $$\frac{2}{n\pi\left(\frac{\omega^2}{\omega_0^2} - 1\right)} \cdot \sin \frac{\omega}{\omega_0}\pi n = 0.7$$

$$\text{or} \quad n = \frac{2.857}{\pi\left(\frac{\omega^2}{\omega_0^2} - 1\right)} \sin \frac{\omega}{\omega_0}\pi n$$

Since $\left|\frac{\omega - \omega_0}{\omega_0}\right| \ll 1$ we may write, as in the working for case 1 above:

$\sin \frac{\omega}{\omega_0}\pi n = \sin \left(\frac{\omega}{\omega_0} - 1\right)n$, giving low values of the argument of the sine, and thus assisting computation.)

Solving the equation graphically we have for the limiting frequencies

n = 7 cycles at 80 c/s or a train of length 0.087 secs.

n = 26 cycles at 300 c/s " " " " " " "

The fact that these times are equal confirms the expectation that for large or moderate integral values of n, i.e. for several complete cycles, the bandwidth is inversely proportional to the number of cycles in the train.

The length of a burst of tone required to excite each filter is thus rather less than the average length of a syllable and it is therefore theoretically impossible to achieve a 10 c/s bandwidth spectrum, since the filters comprising it would have rise and delay times longer than the time of application of the signal.

5.4.2 Possible Forms of the Spectrograph

Fig. 5.4 shows the schematic diagram of a spectrograph of this type.

The twenty-two separate filters are in parallel and receive the amplified signal from the microphone in the studio under investigation. Each filter is followed by amplifying and rectifying circuits, the output of which is a D.C. voltage proportional to the response of the filter. These D.C. voltages are sampled sequentially by a switch, modulated once more to convert into an alternating signal suitable for direct display on an oscillograph or recording on a high speed paper chart recorder. Alternatively,

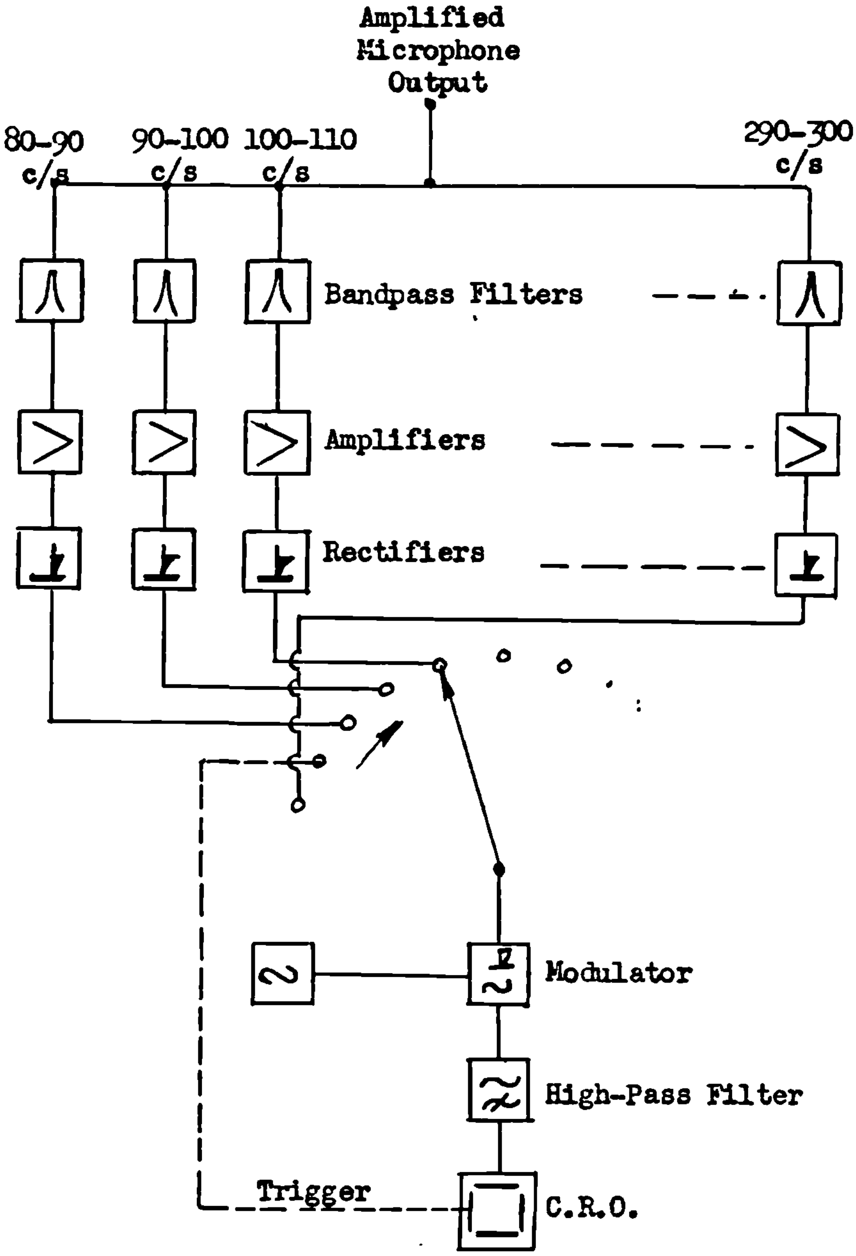


Fig. 5.4 Schematic Diagram of Spectrograph

with the B.B.C. Research Department oscilloscope which has good D.C. stability, the D.C. output of the switch could be displayed directly if it has a suitable voltage range (approximately 0 - 10 volts), an additional contact on the switch providing a trigger pulse to start the timebase sweep at every revolution of the switch.

Each of the main components of this equipment could take several forms which will now be considered. In the next section, the final design of the whole equipment is described as far as it has been finished.

(1) The Filter Sections

The necessity for the construction of 22 component filter sections must have an influence on the design, since unnecessarily complicated design of each section would make the number and cost of components for the whole set prohibitive.

A band pass filter may be a passive network consisting of inductances and capacitors forming resonant circuits of high magnification. This form was not adopted since it would require 44 specially wound inductances at considerable cost.

The alternative possibility is an active filter using capacitors and resistors which can be obtained cheaply and in a range of standard values and combined to make up non-standard values. Alternatively, non-standard values with close tolerances can be obtained at little extra cost. The active types of filter consist of valve or transistor circuits containing frequency-sensitive networks which prevent amplification except in the chosen passband. Active filter circuits using thermionic valves are well-known.

It would have been possible, for instance, to use for each filter a simplified version of the selective amplifier already described, but this was rejected on grounds of expense, bulk, and power supply requirements. It was therefore decided to use transistors as discussed below.

(2) Sampling the Filter Outputs

It is not necessary to scan the whole bank of filters in a time corresponding to the mean time of a syllable of speech, because the reverberation in the room will smooth the fluctuations in each frequency band apart from any redistribution of energy towards the modal frequencies. A display of the outputs of all the filters four times per second is desirable, since the reverberation time of a room may be as low as 0.25 sec. at some frequencies.

Preliminary experiments were made with a rotating switch as used in automatic telephone exchanges. As there were 52 contact positions it was possible to operate the switch at two revolutions only per second, but its operation was electrically noisy and unreliable at these speeds. With some refinements it might be satisfactory but other methods are available.

One method would be to have a series of pairs of valves or transistors arranged as bi-stable multivibrators, so connected that each pair remains in its first stable state until thrown into its second state by a signal from the preceding pair, after which its returns to the first state after an interval determined by circuit time constants, triggering the next pair as it does so. One valve of each pair also acts as a gate connecting one of the filter outputs to the common display circuit.

The method which appears to have the greatest advantages is to use a rotating mirror to scan a series of light-sensitive cells which, when illuminated, provide connections between the filter elements and the display equipment. Until recently, it would have been necessary to use each photo-sensitive cell to actuate a relay or amplifying circuit, since the only available types were either those such as selenium cells which suffered only moderate changes of resistance, or photo-electric types producing small current outputs. Recently, however, cells have been developed such as those using Cadmium Sulphide, in which the resistance changes in the ratio of 10^4 or more upon illumination. Such a cell can be used as a simple conductor between the filter section and the display unit.

5.4.3 The Filter Sections

An experimental filter section was first made according to the circuit of Fig. 5.5 using a single transistor.

The input voltage is applied to the base of the transistor which is connected in the grounded emitter configuration. The collector is connected to the output jack by a 10 μF electrolytic isolating condenser and also through a selective network to the base, thus providing negative feedback. The selective network is of the well-known "parallel-T" type consisting of two T-networks in parallel, one of which has capacitors in the series arm and a resistor in the shunt arm, the other having resistors in the series arms and a capacitor in the shunt arm. The theory of such networks is given, for example, by Punnett (1950).

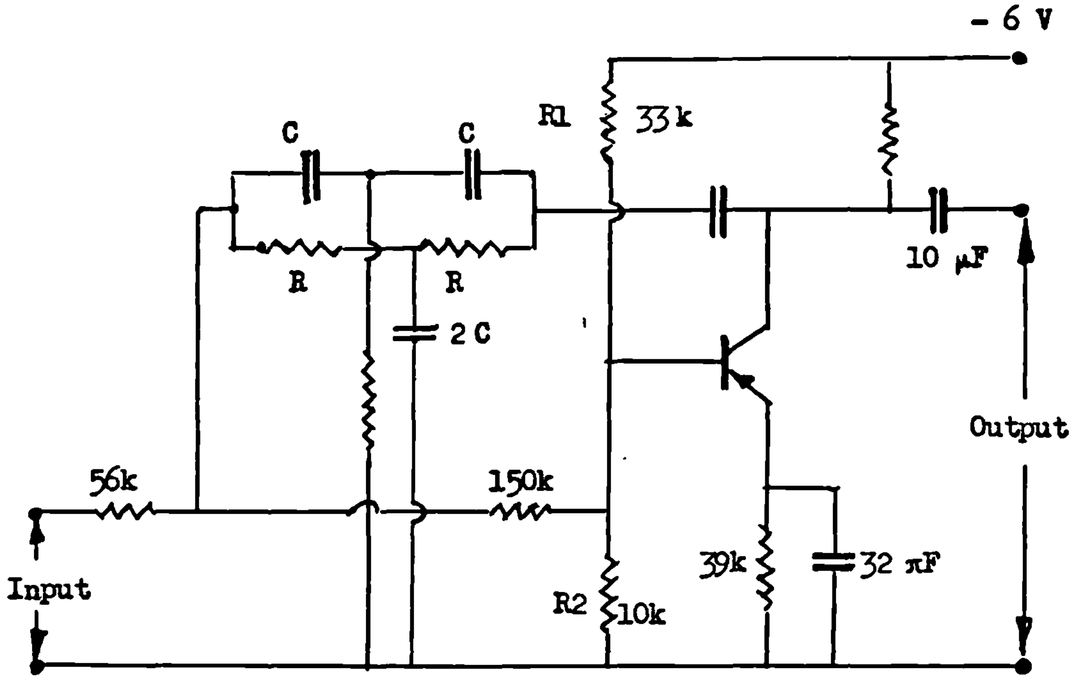


Fig. 5.5 First Experimental Filter

The frequency at which maximum loss occurs in the network is given by $\omega = 1/CR$; the sharpness of the minimum is greatest if the ratios of the C and R value in the two T's are exactly in the ratio two to one as shown in the figure. Fig. 5.6 (a) shows the transmission characteristic of the filter when adjusted to give a minimum of the greatest sharpness. It will be seen that the loss of transmission at the tuned frequency of 80 c/s, compared with that at frequencies well away from it, is of the order of 35 dB, i.e. a ratio of 0.018. At $\pm$ 5 c/s the ratio is 15 dB or 0.18.

Let the current gain of the transistors be μ and the proportion of the output current fed back be β. Let the input and output currents be C_i and C_o respectively.

Then we have $$C_o = \mu(C_i - \beta C_o)$$

Whence gain of stage with feedback $$= \frac{C_o}{C_i} = \frac{\mu}{1 + \beta\mu}$$

A typical value of μ for a transistor stage is 50. Taking β as 0.18 for the edges of the 10 c/s bandwidth we have

$$\frac{\text{Gain at centre}}{\text{Gain at sides}} = \frac{1 + 50 \times 0.18}{1 + 50} = 0.2 \text{ approx. or } 14 \text{ dB.}$$

This selectivity should have been sufficient but tests with the circuit failed to confirm this result, giving the very much flatter curve of Fig. 5.6 (b). Further considerations showed that the loss of selectivity was attributable to the damping of the Parallel-T network by shunt resistances in the circuit, particularly those in the D.C. biasing circuit of the base of the transistor (R_1 and R_2). The biasing circuit was modified, increasing

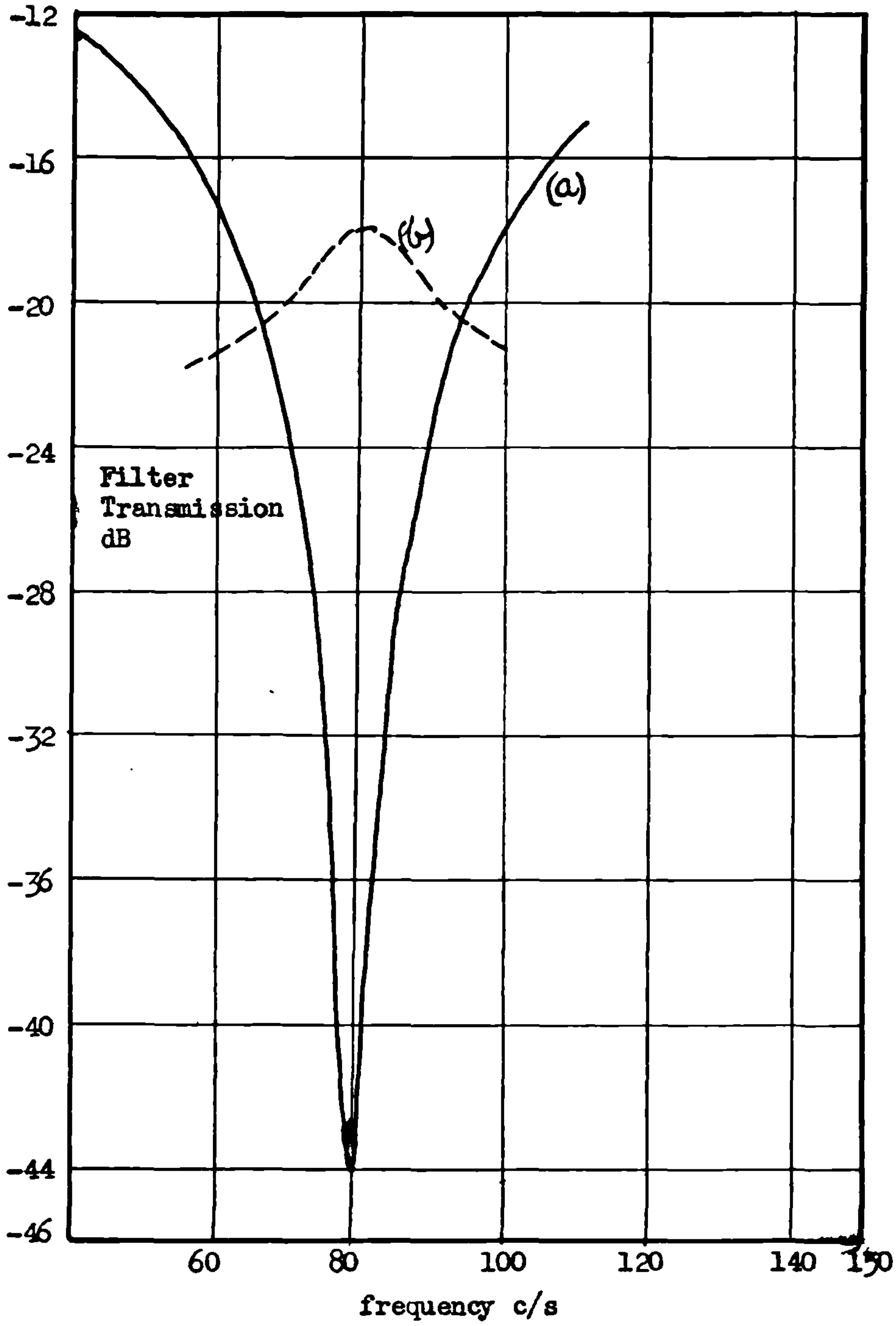


Fig. 5.6 Transmission Characteristics of First Filter

(a) Parallel-T Network
(b) Amplifier Stage with Feedback

resistance values, but there was little improvement. It was concluded that the effect was partly due to the conductance of the transistor itself, through the emitter and collector, and though higher impedance transistors were tried with some success, it was decided to redesign the circuit, using positive feedback through a selective network.

Fig. 5.7 (a) shows the final circuit designed by the author's colleague K.F.L. Lansdowne. This consists of two identical circuits in series, transistors VT1, 2 and 3 being in the first, and VT4, 5 and 6 in the second.

VT1 is biased by resistors R2, R3 and R4, and an output is taken across R4, the emitter resistor which has no capacitor in parallel. VT2 and VT3 form the amplifying stages with negative feedback through R11 and the preset variable R15, to improve linearity and allow the gain to be adjusted. The base of VT2 receives a signal from VT1 through R5 and C2 in parallel, and also a signal from the collector of VT3 fed back through R6 and C3 in series. R5, R6, C2, C3 together form the frequency-selective network, R5 being equal to R6 and C2 to C3.

The admittance of R5, C2 is given by

$$\left(\frac{1}{R} + j\omega C\right) = \frac{1 + j\omega CR}{R} \qquad \text{where } \begin{array}{l} R = R5 = R6 \\ C = C2 = C3 \end{array}$$

The admittance of R6, C3 is similarly

$$\frac{1}{R + \frac{1}{j\omega C}} = \frac{\omega^2 C^2 R^2 + j\omega CR}{\omega^2 C^2 R^2 + 1}$$

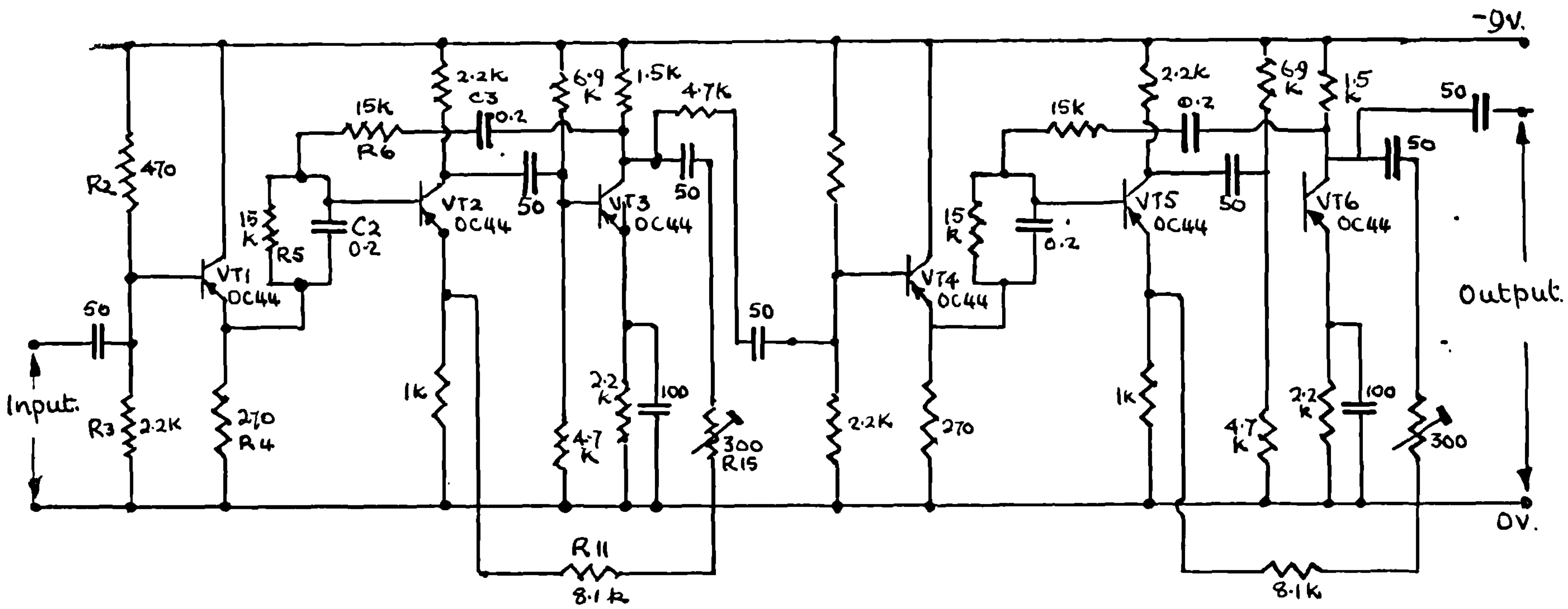


Fig. 5.7 Selective Amplifier Circuit Diagram for A.F. Spectrometer.

Hence the phase angles of the component currents at the input to VT2 are

$$\tan^{-1}\omega CR \quad \text{and} \quad \tan^{-1}\frac{1}{\omega CR}$$

If $\omega = 1/CR$ these two are equal and positive feedback takes place giving any desired gain at this frequency up to actual instability, according to the gains of the forward and feedback loops, which may be adjusted by R15.

The second identical circuit in series with the first part described was introduced to obtain the best compromise between bandwidth and rate of decay of the filter. The single circuit gave the correct bandwidth only if the negative feedback was reduced to a point where the decay rate of the circuit was excessive, having a value of approximately 0.5 sec. By adding a second circuit in series the bandwidth was reduced, enabling the feedback to be increased and attaining a decay time of 0.33 sec. comparable with the shortest reverberation time likely to be encountered, at any rate in an operational talks studio. Fig. 5.8 shows the frequency characteristics of a single circuit and the two in series, with the decay times marked.

There is an apparent contradiction in this independent variation of decay time and bandwidth; in the preliminary calculations to ascertain the feasibility of the construction of these filters (see section above) the number of cycles necessary to establish a frequency with a predetermined bandwidth of uncertainity was seen to be approximately inversely proportional to the bandwidth.

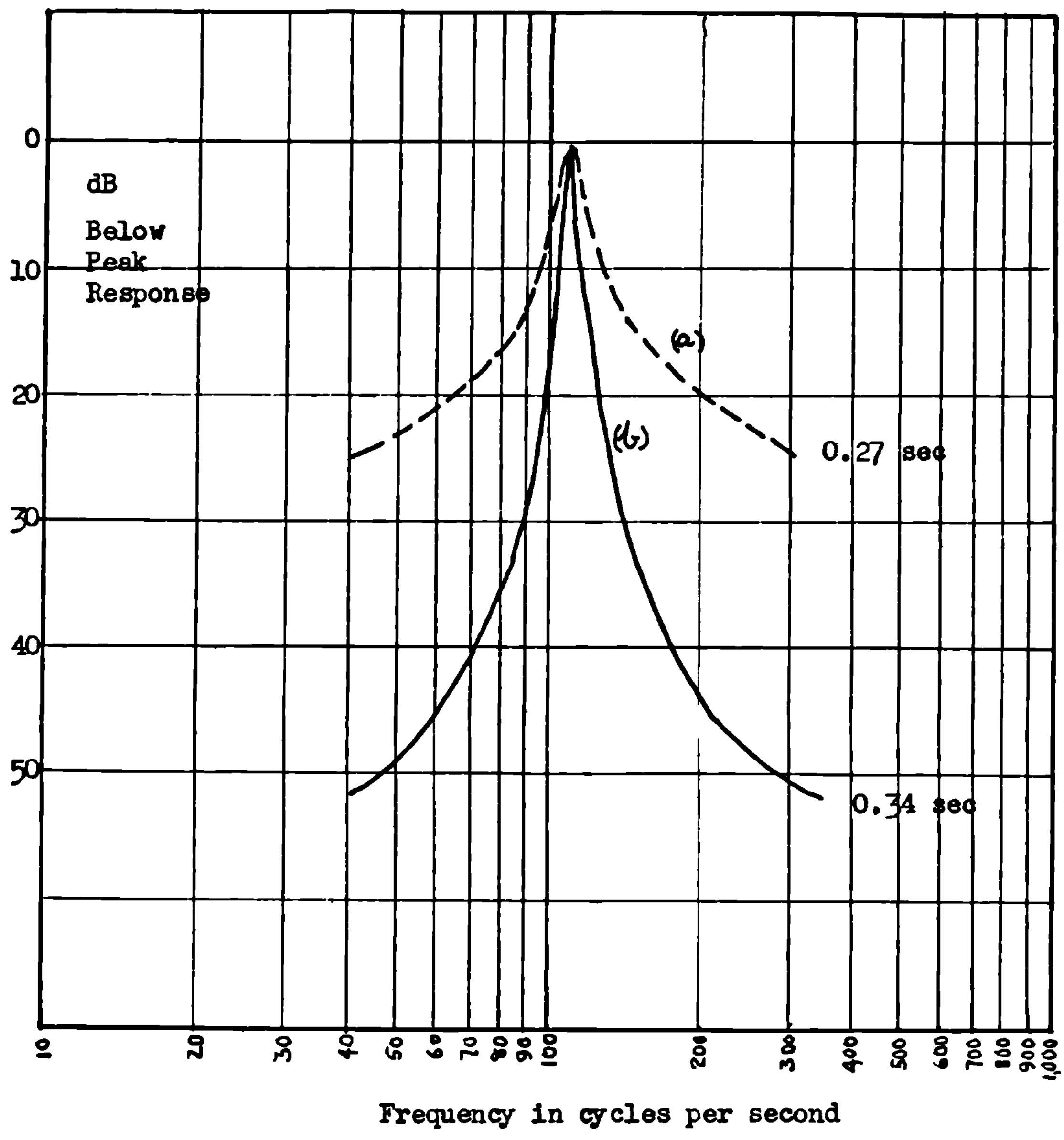


Fig. 5.8 Response of Filter Section

(a) One Circuit: Decay time 0.27 sec.

(b) Two Circuits making Complete Filter: Decay time 0.34 sec.

Similarly, if the filter is regarded as a "black box" having a transfer function of a certain bandwidth, and ultimate rejection, its resemblance to a simple tuned circuit is evident and it is tempting to suppose that similar relationships must hold between bandwidth and decay time as are found with simple tuned circuits. Moreover, with a single active R.C. circuit such as that comprising VT1 transistors, VT2 and VT3, the decay time is in fact approximately proportional to the reciprocal of the bandwidth.

On the other hand, it is clear that, provided there is suitable buffering between two circuits placed in series, the first of the two will be unaffected by the presence of the second. The collector of VT3 is a low-impedance point, due to the heavy negative feedback across the two stages, provided by C3, R6, R15 and R11. R14 and the high input impedance of VT4, further increased by the negative feedback arising from the undecoupled emitter bias resistor R22, therefore effectively isolate the first circuit from any effect of the second. The second circuit therefore receives as an input precisely the same time function as would a passive resistive load put in its place, and modifies it in the same way as it would modify an input from a passive source.

Thus with two circuits in series the pass band width would be reduced. The amount of the reduction would depend on the shape of the frequency characteristic at the peak. If the response falls on either side of the peak linearly with frequency deviation from the peak frequency, the ratio of the response at $\pm$ 5 c/s to that at peak will be squared by the addition of the second circuit, i.e. the number of decibels down from peak will be doubled. Similarly the deviation for a given reduction in response (say 3 dB) will be halved.

The simpler relationship giving a symmetrical peak with zero gradient is a square law between dB reduction and deviation. For two circuits, the bandwidth will be that between the frequencies at which the single circuit gives a reduction $1\frac{1}{2}$ dB, i.e. $1/\sqrt{2}$ of the single-circuit bandwidth. It is difficult to measure the exact shape of the peaks region of the curve but examination of Fig. 5.8 suggests that the actual law is nearer to a linear fall-off than to a square law.

Following the reasoning on the behaviour of independent circuits in series, as given above, we may attempt to calculate the change in decay time as measured by the methods described in Section 7.2 below for the reverberation times of rooms. The second circuit is initially forced into oscillation by an input the amplitude of which we shall represent by Vo, i.e. the amplitude of the first circuit, and does not start to decay until the forcing oscillation is reduced. We assume that the rate of decay is then proportional to the difference between its own amplitude V and the instantaneous value Vb of the forcing amplitude applied to it.

Thus we have $\frac{dV}{dt} = -k(V - Vb)$

and $Vb = Vo\ e^{-kt}$ (first circuit performing a classical damped oscillation with a rate of decay proportional to the instantaneous amplitude.)

Hence, $\frac{dV}{dt} = -k(V - Vo\ e^{-kt})$

Rearranging, $\frac{dV}{dt} + kV = kVo\ e^{-kt}$

Multiplying by e^{kt} thus gives

$$\frac{d}{dt}\ (Ve^{kt}) = kVo$$

and hence integrating

$$\left[Ve^{kt}\right]_o^t = \left[Ve^{kt}\right]_o^t$$

$$\therefore\ Ve^{kt} - Vo = kVt_o$$

and finally $V = Vo(1 + kt\ e^{-kt}$ or $\frac{V}{Vo} = (1 + kt)e^{-kt}$

The measured decay time for the single circuit was 0.27 sec. for a reduction of 60 dB, i.e. to one thousandth of its original amplitude.

This gives $k = \log_e 1000/0.27 = 25.5$, and substituting this value into the equation for V/Vo we get Fig. 5.9 (b). Curve (a) is the function $V/Vo = e^{-kt}$, the decay curve for a single circuit.

The method of measurement of reverberation time is simulated by the straight line curve (c), drawn as a best fit over a part of the curve (a) between 5 dB and 35 dB below the starting level, as specified in British Standard 3638 (1963). The slope of this curve corresponds to a decay time of 0.364 sec., in close agreement with the measured time of 0.35 sec. for the two circuits in series.

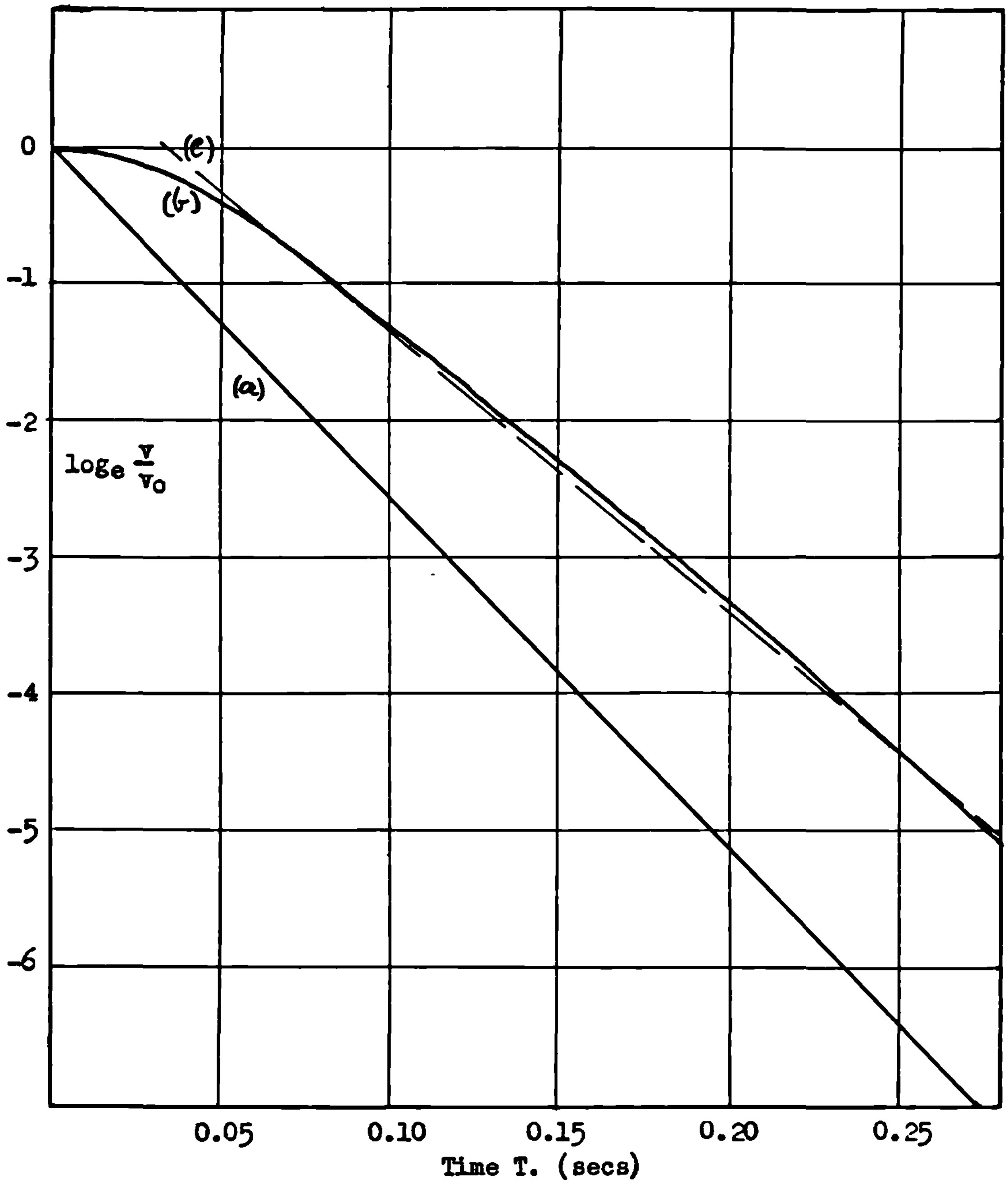


Fig. 5.9 Free Decay Times of Single Band-Pass Filter and Two Identical Filters in Series

(a) Single Circuit
(b) Two Identical Circuits
(c) Best-Fit Straight Line to (b)

It is concluded that the two circuits are in fact operating independently. The combination gives satisfactory results with respect to bandwidth and decay time, and the filter design may be regarded as adequate for the intended purpose.

Fig. 5.10 is a photograph of one of the two-circuit filters which are built on printed-circuit boards. The full set will cover every 10 c/s interval from 80 c/s to 300 c/s.

5.4.4 The Display Equipment

At the time of writing it has not been possible to complete the spectrograph, and this section describes briefly the display equipment which has been designed. Some parts of it have been tested in "breadboard" form; the switch, however, has not been started.

Fig. 5.11 shows the scheme for the display equipment. The first stage is a full-wave rectifier and smoothing ciruit, the purpose of which is to enable the switch to sample the envelope of the output of each filter. If we assume a scanning rate of 150 filters per second, each filter will be sampled for less than 1/150 sec. and consequently a display of the instantaneous output would show less than one cycle of the output of each filter less than 150 c/s in frequency. It is therefore necessary to convert the sinusoidal output of each filter into a rectified and smoothed signal, the time-constant of the rectification being of the order of the interval of time between successive samplings of a particular filter.

The filter output is fed through the 50 μF capacitor

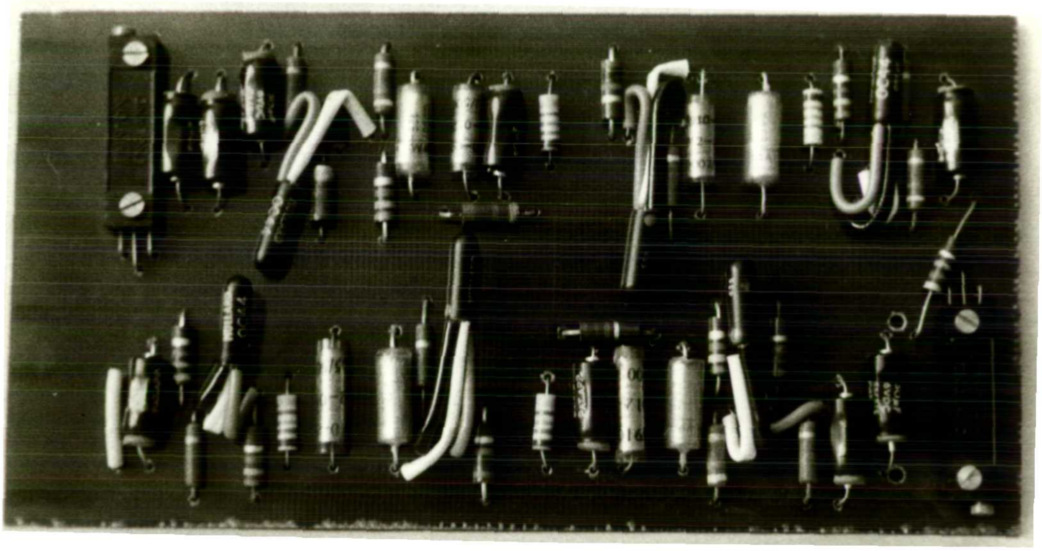


Fig. 5.10 · Photograph of Printed-Circuit Board Carrying Complete Filter.

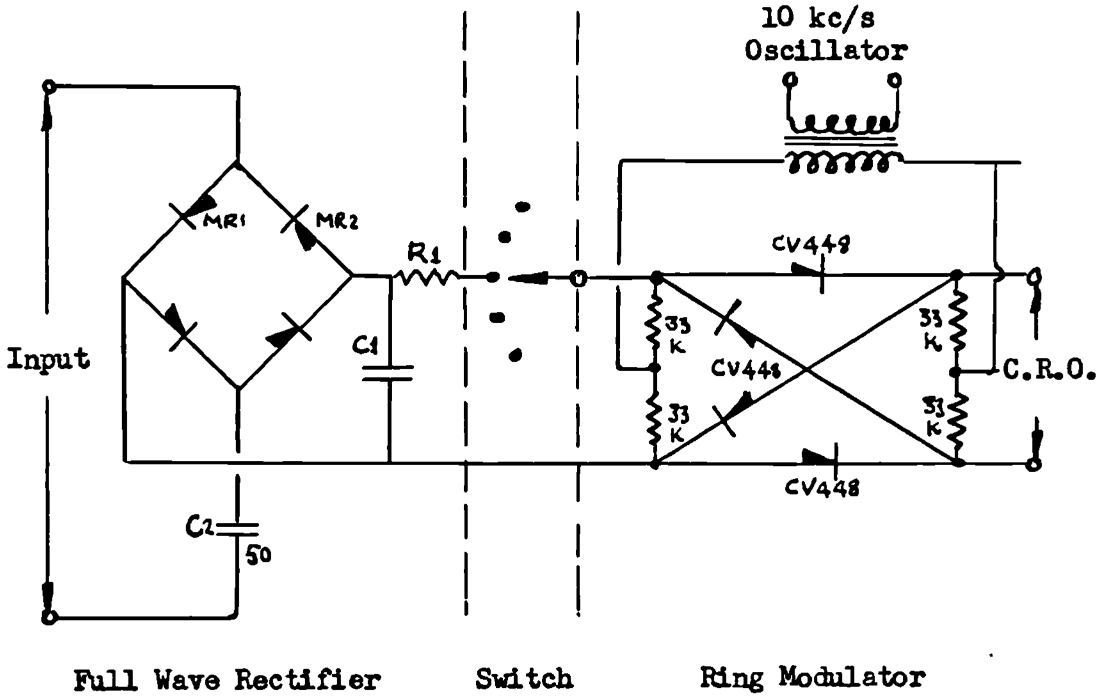


Fig. 5.11 The Display Equipment

(Fig. 5.7) to a bridge rectifier consisting of four low-impedance germanium diodes such as the GEX 64. The 50 μF capacitor C2 isolates the circuit from earth, enabling the rectifier which follows to work without an input transformer. The network C1 and R1 determines the time constant of the circuit; with C1 = 0.05 μF and R1 = 6K this would be 0.3 sec.

The switch design is shown in Fig. 5.12. For completion of the circuits it uses the fall in resistance of a cadmium sulphide cell, one to each filter, this possibility having been suggested by K.F.L. Lansdowne. The cells are arranged in a ring, successive cells being placed on opposite sides of the ring to reduce the waiting time between scanning two successive cells as will be explained below. The axle of a synchronous electric motor performing five revolutions per second passes through the axis of this ring and to it is attached a scanning box which consists of a short cylinder of thin metal sheet with holes in the two flat faces through which light can pass to the cadmium sulphide cells. Above, and concentric with, the scanning box is a lamphouse containing a bulb of approximately 60 watts. The two holes in the lower face of the scanning box are diametrically opposite to one another and on the same centre-circle as the cells. Their radial width is just sufficient to span the light-sensitive element but they are elongated circumferentially to the length of the spacing between two cells. The upper face is pierced with two holes with radial side edges. The leading edge of each hole is at a distance behind the corresponding edge of the hole in the lower face equal to the width of the

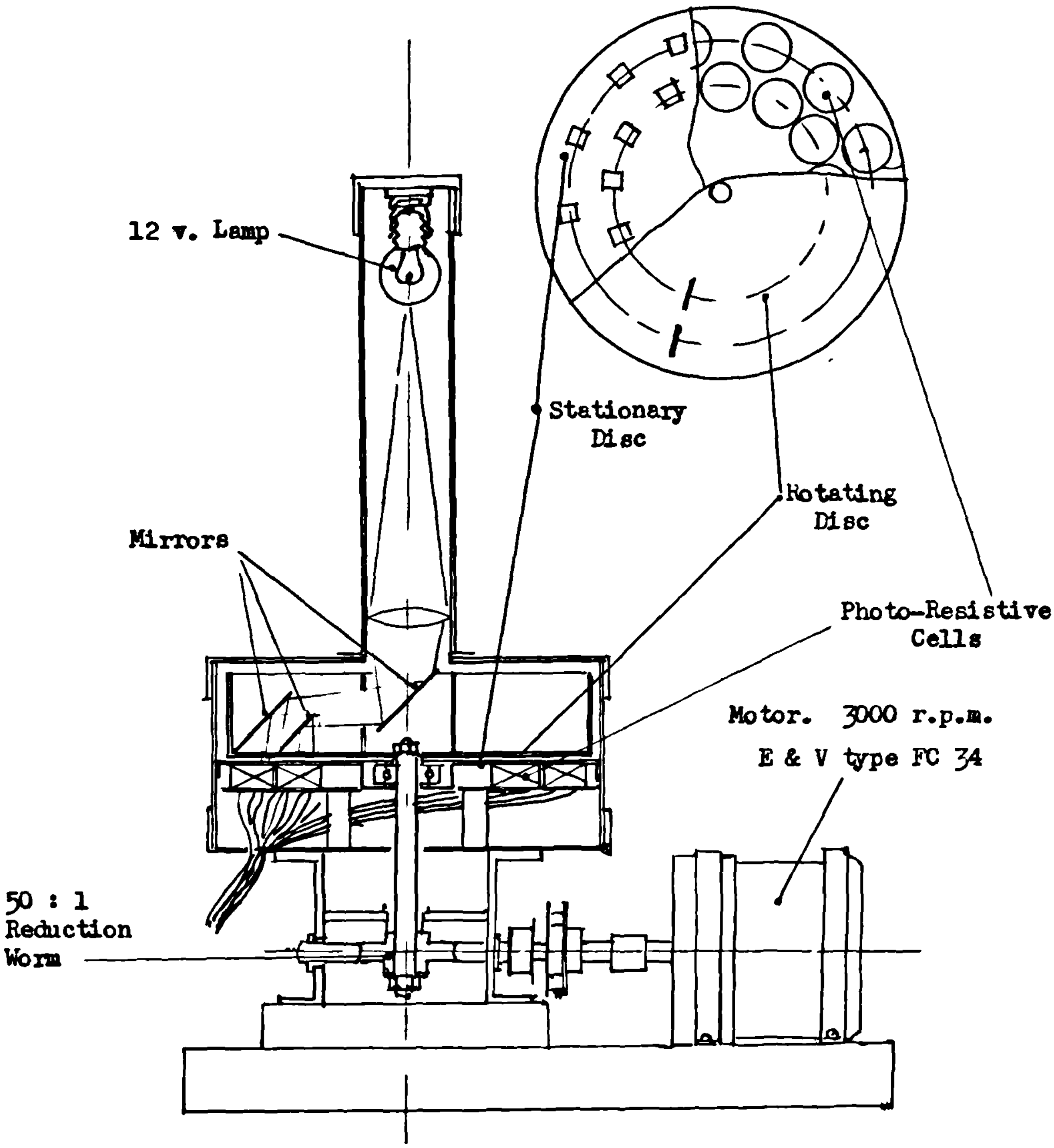


Fig. 5.12. 22-way Rotary Photo-Resistive Switch (Scale ½)

light-sensitive element. There are two radial slits in the stationary lamphouse opposite to the sector holes in the upper face of the scanning box, so that as the box revolves light from each slit passes for a short period twice per revolution into the scanning box. The upper face of the box is lined with opal perspex so that light entering through the upper holes is opposite to the trailing edge of the lower holes. The spacing between the cells is made just over twice the width of the light-sensitive part of the cell. The second cell in the sequence is in the opposite half of the circle and staggered with respect to the first so that its centre is radially opposite to a point halfway between the first and third cells.

In operation, therefore, the cells are illuminated sequentially with small intervals or overlapping periods which may be controlled by the lengths of the illuminating holes and the width of the slit. The details of the design are worked out for a slight overlap during which the illumination of one cell is falling as that on the next is rising at the same rate.

Light barriers, and matt black or white paints, are used to ensure that the cells are as fully illuminated as possible during the 'on' periods and as dark as possible when off.

The switch is followed by a conventional ring modulator in which the d.c. output of the switch modulates a sinusoid of high enough frequency to give a trace free from ripple on a rectified display c.r.o. or high-speed level recorder of specified time constant.

Construction of this equipment is proceeding fast but it is feared that it will not be complete enough for a trial before this thesis goes to press. In view of the uncertain and unsatisfactory nature of other tests for colourations given in this and the previous section, considerable importance attaches to the success of the spectrograph method, since the objective investigation of such an evanescent subjective phenomenon as colourations of speech depends vitally on reliable assessments of the severity of the colourations, and the frequencies at which they occur, and has met continued difficulties in establishing cross-correlations.

CHAPTER 6

THE USE OF SELECTIVE ABSORPTION FOR THE SUPPRESSION OF UNWANTED MODES

6.1 General

It has been shown that subjectively appreciated faults in a small room are associated with strong isolated modes or groups of modes separated from their neighbours by an interval of 20 c/s or more on each side. To remedy the defect by absorption it is necessary to consider carefully the bandwidth of the absorber. If it is much larger than the interval between adjacent axial modes it will partially absorb sound energy composing a large number of modes without altering their relative excitation or spacing, or in consequence their subjective effects. If, on the other hand, the bandwidth of the absorber is slightly greater than the relevant mode spacing, it will reduce the excitation of a few modes and might possibly produce worse colouration at an adjacent frequency.

The bandwidth of a resonant absorber to be used in a small room, particularly a broadcasting studio with a comparatively short reverberation time, has a lower limit, determined by its own decay time which should not exceed the reverberation time of the room since in this event it would continue to re-radiate sound after the sound field in the room has ceased to exist.

If we express the equation of motion of any resonant system in free decay as:

$$m\ddot{x} + R\dot{x} + \zeta x = 0$$

x = displacement
m = mass
R = Resistance coefft. per unit velocity
ζ = Restoring force constant.

Then the solution is (e.g. Davis 1934)

$$x = x_o e^{-Rt/2m} \cos(\omega t + \delta)$$

where δ is a phase angle,

$$\omega_o^2 = \gamma/m.$$

If R/m is small the angular frequency $2\pi f$ is approximately equal to $\sqrt{\gamma/m}$.

Writing X as the amplitude at a time τ, at which the amplitude has decayed to one thousandth of its starting value, we have, $X/X_o = 1/1000 = 1/e^{+R\tau/2m}$.

Whence the reverberation time $\tau = \frac{2m}{R} \log_e(1000)$

$= 13.8\ m/R$

$= 13.8\ Q/\omega \qquad Q = $ Q-factor

$= \omega m/R.$

Referring again to simple theory of resonators, the bandwidth is defined as the frequency interval between the points where the amplitude falls to $1/\sqrt{2}$ of its value at resonance and is given by $B = f/Q$ where f is the frequency at resonance.

Hence

$$\tau = \frac{\log_e(1000)}{\pi B} = \frac{2.20}{B} \qquad \text{.........} (6.1)$$

This shows that for a given reverberation time, the bandwidth in cycles per second is not adjustable and is independent of the frequency of resonance.

It is necessary to consider also a related phenomenon. Rschevkin (1938) has shown that a resonator of high magnification, Q, adds an effective volume equal to Q^2 times the volume V' of its cavity to that of the room. At the same time it supplies damping equivalent to an absorbing area A', which he evaluates.

The reverberation time of a room of volume V and total absorption A is then given by a modified form of Sabine's (1922) equation, viz.:

$$T = \frac{0.049\ (V + Q^2 V')}{A + A'}$$

where A is the total absorption of the surfaces of the room.

If $\frac{Q^2 V'}{A'} > \frac{V}{A}$ the reverberation time of the room-resonator system will be greater than that of the room alone, and this is shown by Fig. 6.1 which gives the results of reverberation time measurements of a small studio. Curve (a) is the reverberation characteristic of the room alone, and (b) is the result of adding resonators of high Q.

This result will be applicable to resonant absorbers of any kind since they necessarily store a certain amount of energy when excited by a sound field and may give back the energy over a larger effective period than the reverberation time of the room. Possible types of absorber will be considered in the following sections.

The basis for the design of resonant absorbers has been given in two papers by the author (1952, 1952-3). The essential parameters are the frequency of resonance and the internal resistance of the absorber which must be made approximately equal to the radiation resistance of the sound field to the absorber. Bandwidth must also be considered, to ensure that the absorber works over a

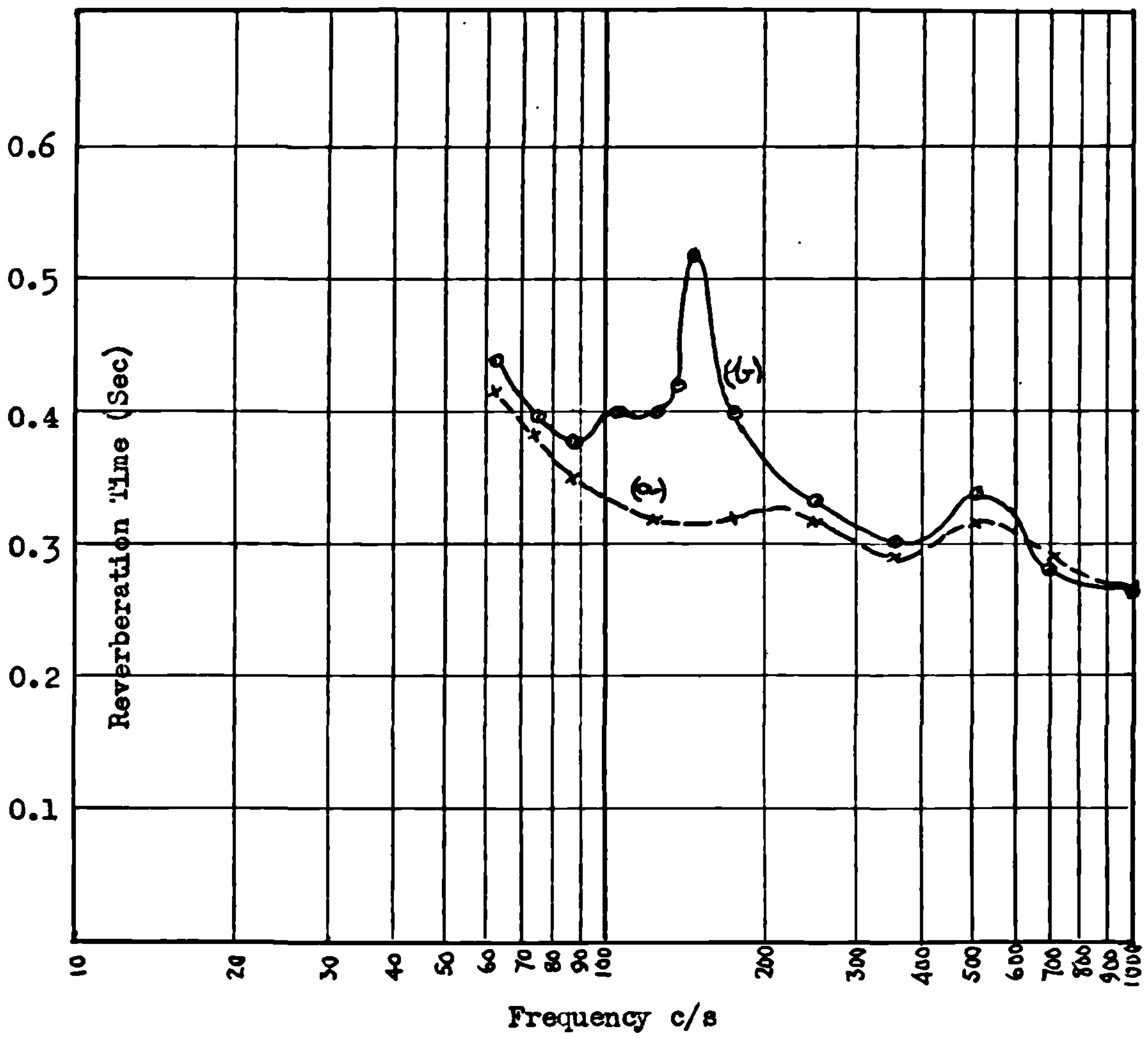


Fig. 6.1 Effect of high-Q Resonators on Reverberation Characteristic of Small Studio

(a) Without Resonators
(b) With 84 Resonators tuned to 140 c/s

suitably restricted or extensive range of frequency.

Fig. 6.2 shows a circuit analogous to a simple absorber in which all the elements of the acoustic input impedance can be considered as lumped constants, i.e. an absorber which is small compared with the wavelength of sound at the frequency considered. R_r, X_r are the resistive and reactive components of the radiation resistance, while R_f, X_f refer to the internal impedance of the absorber.

The power dissipated in the absorber (R_f, X_f) is given by

$$\frac{P^2 R_f}{(R_r + R_f)^2 + X^2} \qquad \text{................ (6.2)}$$

and if X is small compared with $(R_r + R_f)$ as in a deep layer of resistive material or in a resonant system at its frequency of resonance, the dissipation is

$$\frac{P^2 \; R_f}{(R_r + R_f)^2} \qquad \text{................ (6.3)}$$

giving a maximum value of $P^2/4R_r$ where $R_f = R_r$, i.e. the correctly matched condition.

For unit area of plane-wave sound field, $R_r = \rho c$ where ρ is the density of air and c is the velocity of sound; hence the dissipation is $P^2/4\rho c$ for unit area of an infinite plane sheet of a perfect absorber.

The ratio of the maximum possible absorption by any other absorbing body with radiation resistance R_r to that of unit area of

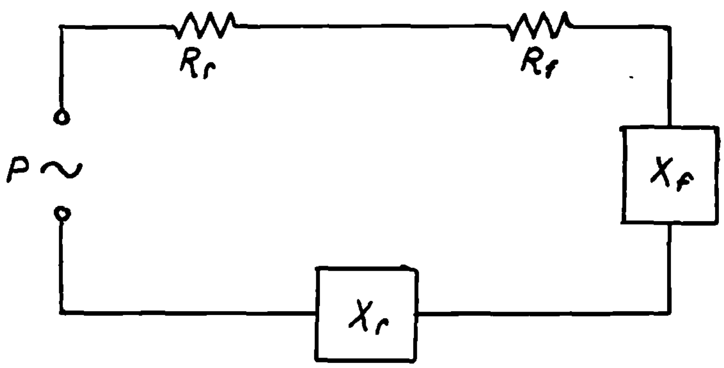


Fig. 6.2 Circuit Analogous to Absorber

P = Sound Pressure
R_r = Radiation Resistance
R_f = Internal Resistance
X_r = Radiation Reactance
X_f = Internal Reactance

perfect absorber is therefore $\rho c/R_r$; .... (6.4)
in other words, this ratio represents the number of units of area of a perfectly absorbing plane surface to which the body is equivalent, provided that the internal or dissipative resistance of the body is correctly matched to the radiation resistance. That is to say, this ratio shows its maximum possible absorption in a diffuse field in terms of what Sabine (1922) called "open window units".

In this chapter the absorption of a body, expressed as the equivalent area of a perfectly absorbing plane surface will be termed the absorbing cross-section; the maximum possible value, assuming a state of resonance and correct matching, as given by expression (6.4), will be termed the maximum, or maximum possible, absorbing cross-section.

The value of R_r for a small hole in an infinite plane wall may be shown to be: (Rayleigh, 1878, Vol. 2, p. 319)

$$2\pi\rho c/\lambda^2$$

where λ is the wavelength of the sound, ρ is the density of air and c is the velocity of sound.

Hence the maximum absorbing cross-section is

$$\lambda^2/2\pi$$

These results will be used in the consideration of Helmholtz, membrane and functional absorbers in this and the next chapter.

6.2 Helmholtz Resonator Absorbers

A considerable amount of literature already exists on the use of Helmholtz resonators for absorption, e.g. Gilford (1952), Zwikker and Kosten (1949) and Bruel (1951). The Helmholtz resonator consists of a volume of air contained in a rigid enclosure, communicating with the air outside through a hole, slot or tube. The air in the passage or neck between the enclosed space and the outer air acts as a mechanical or acoustical inertance while the enclosed air has a compliance which is proportional to its volume. The system has a resonance, therefore, which is determined by these two components; at the frequency of resonance, the velocity of air flow through the neck reaches a maximum, and the resistance of the neck will absorb energy, abstracting it from the incident sound. If the resonator is carefully designed and constructed, using rigid smooth materials throughout, the resonance may have a Q-factor as high as 40. From this figure downwards, the Q-factor may be adjusted to any amount by inserting extra resistive materials in the neck. Ward (1952) describes the adjustment of resonators in this way, using gauzes as the resistive materials. The frequency of resonance may be varied over a wide range by suitable design, both the volume and the length and cross-section of the neck being independently variable. There are various limitations to the design of resonators for a particular purpose. In the first place, the shorter the neck the greater the conductivity; but the effective length of the neck is greater than its actual length by an amount which has been given by Rayleigh (1878, p. 171) as 1.7

times the radius of the neck if circular. Thus, although a wide short neck in conjunction with a specified volume gives a high frequency, increasing the diameter of the neck eventually ceases the raise the frequency because the effective length of the neck starts to increase almost linearly with the diameter. Secondly, there is a lower limit to the volume of a resonator, which will be effective in a particular application; assuming that the frequency of maximum absorption has been decided, this frequency can be obtained for a wide range of corresponding volumes and neck conductivity. If the volume is decreased, the conductivity of the neck must be reduced to maintain the same frequency and hence the Q-factor increases unless the resistance of the neck is increased in the same proportion. But the resistance must be related to the radiation resistance of the sound field, being equal to it if the maximum absorption at resonance is required, or somewhat larger to attain a wider bandwidth without seriously reducing the absorption at the resonance.

Thus if the volume is too small, the result will be either a good absorption over too narrow a frequency band, or a poor absorption at all frequencies.

Van Leeuwen (1962) has recently published an excellent account of the theory of absorption by Helmholtz resonators in a small room, with special reference to the damping of specific modes. He shows that a room-resonator combination may be represented by an analogous electrical circuit in which the room at a modal frequency f_n is represented by an inductance M_n, a capacitance $\frac{1}{s_n}$ and a resistance in parallel, and across these are the resonator constants M_r, R_r, $\frac{1}{s_r}$ in series with one another. For simplicity we are here considering the case where the resonator is at an antinode of the mode and therefore receives the maximum excitation. Van Leeuwen calculates the values of these six constants in terms of the reverberation time of the room τ_n, the frequency f_n, the Q-factor of the room Q_n and of the resonator, Q_r and the volumes of the room and the resonator V , V_r. The resonator is assumed to have been tuned to the eigenfrequency in question.

He shows that the coupling factor between the two circuits is given by:

$k = \sqrt{\frac{A_n V_r}{V}}$ provided always that the resonator is at an antinode, where A_n is a constant of value 2, 4, 8, according to whether the mode is axial, tangential or oblique. As we are mainly concerned with isolated axial modes, we may write:

$$k = \sqrt{\frac{2V_r}{V}}$$

The paper proceeds with the calculation of the reductions in reverberation time and steady-state sound level due to the presence of the resonator and gives a design chart to enable a resonator giving any particular reductions to be designed. Van Leeuwen finds that the most favourable condition is that in which both quantities are reduced in the same ratio, and for this he gives the following equations:

$$\text{Reduction factor} \approx 1 + 0.3k\, fn\tau_n$$

$$Q_r \approx 0.65/k$$

These formulae are valid for reduction factors greater than 2, except that the stated reduction in τ is not achieved if the neck is very short in comparison with its diameter, i.e. if the Rayleigh end correction is comparable with the length of the neck.

Van Leeuwen quotes experimental tests of these results, but unfortunately confines himself to experiments in a room of 41 m^3 (1,450 ft^3) volume which prior to treatment had a reverberation time of approximately 4 sec. at a modal frequency of 84 c/s. Adding a resonator of 37 dm^3 (1.3 ft^3) volume and a Q-factor of 41 reduced the reverberation time to 2.2 sec. Increased damping making Q_r equal to 13 reduced this to 0.8 sec. The bandwidth over which this reduction took place was approximately 4 c/s.

Using Sabine's formula (Sabine 1922) relating the reverberation time of a room and the total absorption* Σ,

$$\text{viz.} \quad = \frac{0.049\ V}{\Sigma} \qquad (ft^2)$$

$$\text{We have} \quad A_r = 0.049 \left\{\frac{1}{0.8} - \frac{1}{4.0}\right\} \times 1450 \quad ft^2$$

$$= 71\ ft^2$$

---

* In Chapter 8 below, Eyring's formula is used in preference to Sabine's for reasons there given. Here, not knowing the surface area of the room, it is not possible to use Eyring's formula and Sabine's - which is a good approximation in lightly damped rooms - is therefore used instead.

Experiments by the present author have been almost exclusively confined to talks studios, in which the reverberation time inthe region of the colouration frequencies was 0.6 sec. or less.

Let us calculate, by van Leeuwen's methods, the reduction obtainable from a resonator of the same size as he used, starting from a reverberation time of 0.6 sec. Here $V_r = 1.3$ ft^3, $V = 1450$ ft^3,

$$\text{Then } k = \frac{1.3 \times 2}{1450} = 0.042$$

And reduction factor $= 1.67$.

This gives a value of 0.36 for the reverberation time of the mode, which would be a suitable value for the reverberation time of the studio in the region of the mode.

The reduction here predicted from 0.6 secs. to 0.36 sec. reverberation time is equivalent to an increase in total absorption of

$$0.049 \times 1450 \times \frac{1}{0.36} - \frac{1}{0.6}$$

$= 78$ ft^2 which is rather greater than that achieved in van Leeuwen's experiments.

As shown in Section 6.1 above, the theoretical maximum absorption cross-section of a resonator is $\lambda^2/2\pi$.

At 84 c/s this equals 29 ft^2 which is very much less than that found by van Leeuwen.

The maximum value of $\lambda^2/2\pi$ is, however, for a resonator in a diffuse sound field, and would be equivalent to the average obtained by sampling a stationary wave system equally in all parts of the room. Van Leeuwen's calculations, being for a resonator placed accurately at an antinode of the system will give higher values. The mean value for all parts of the standing wave field would be lower than the maximum by a factor equal to the mean square of the pressure divided by the maximum pressure squared. Since the field is sinusoidal in form, this factor is one half, which reduces the maximum absorption predicted by his methods to a figure of the same order as $\lambda^2/2\pi$.

Exact agreement would not be expected since the reverberation formulae are statistical approximations. In the case considered, the surface area of the room would probably be of the order of 800 ft^2. The total absorption of 196 ft^2 for the reverberation time of 0.36 sec. would therefore represent a mean absorption coefficient of 0.25. For so high a mean coefficient it would be more accurate to use Eyring's formula than Sabine's formula in calculating the total absorption. However, the absorption given by Eyring's formula depends on the surface area of the room and not only upon the volume, and the surface area will vary independently of the volume. Van Leeuwen's calculations include only the volume of the room; the dimensions affect only the frequencies of the modes and cannot therefore agree in general with the results of Eyring's formula, though these are usually more accurate than those of Sabine's formula.

The application of this work depends upon:

(1) The identification of a mode causing a colouration,

(2) The exact measurement of its frequency and determination of antinodal lines or planes.

In practical circumstances this has usually proved difficult since the initial treatment of talks studios invariably reduces the reverberation time to the order of 0.4 to 0.5 secs., and the ratio of reverberant to direct sound when the studio is excited with tone from a loudspeaker is comparatively small. To plot the standing wave systems, steady-state methods have been used, in which a loudspeaker is set to emit steady tone at the modal frequency, and a probe microphone samples the room space to find positions of maximum and minimum sound pressure.

Resonators made for damping of modes must be carefully made. They must be extremely rigid to avoid loss of energy by vibration of the sides and all cracks must be carefully filled to avoid forming resistive shunt paths.

It should be possible to tune the resonators to any frequency within a band enclosing the modal frequency. Ward (1952) described one method of tuning resonators in which the necks are cylindrical, by providing cardboard tubes which may be slid into each other to vary the length of the neck.

More recently resonators in which the neck is a long slit have been tried for selective absorption. The theory of these has been given by Pedersen (1940). They have higher inherent resistance and are therefore not adjustable over such a wide range of resistance values as the cylindrical-neck resonators, but they may be constructed to allow easy adjustment of frequency by alteration of the slit width.

Fig. 6.3 shows a design in which the width of the slit may be continuously varied for tuning. It is designed to span a corner of a room, which of course is an antinodal pressure region for all modes. The adjustment is effected by screwing wood-screws which pass through a lath of wedge section into the transverse supporting beams. Pieces of rubber tubing fitting into recesses in the supporting beams act as return springs to hold the lath in place. There are two slots in parallel which behave as one slot of twice the width except that there is a smaller end-correction to the length of the slot and the undamped resistance is somewhat greater than for a single slot.

The length of the slots is pre-determined by the construction of the two shaped fillets into which the lath fits. Resistive material (fine metal gauze or cloth) can be fixed in the bottom of the recess or in the shaped fillets.

The advantages of corner mounting for resonators are

(1) The corner is at a pressure antinode for all modes

(2) The corner space is seldom required for other purposes

(3) The resonator does not project into the room, even though the main volume is not recessed behind the wall surfaces.

For this reason spaces in the corners are being reserved in all small studios now being planned for the B.B.C.'s regional head-quarters in Cardiff and elsewhere. The walls of these studios are to be treated with a modular construction built in a wooden frame-work 20 cm deep which covers all parts of the walls except those

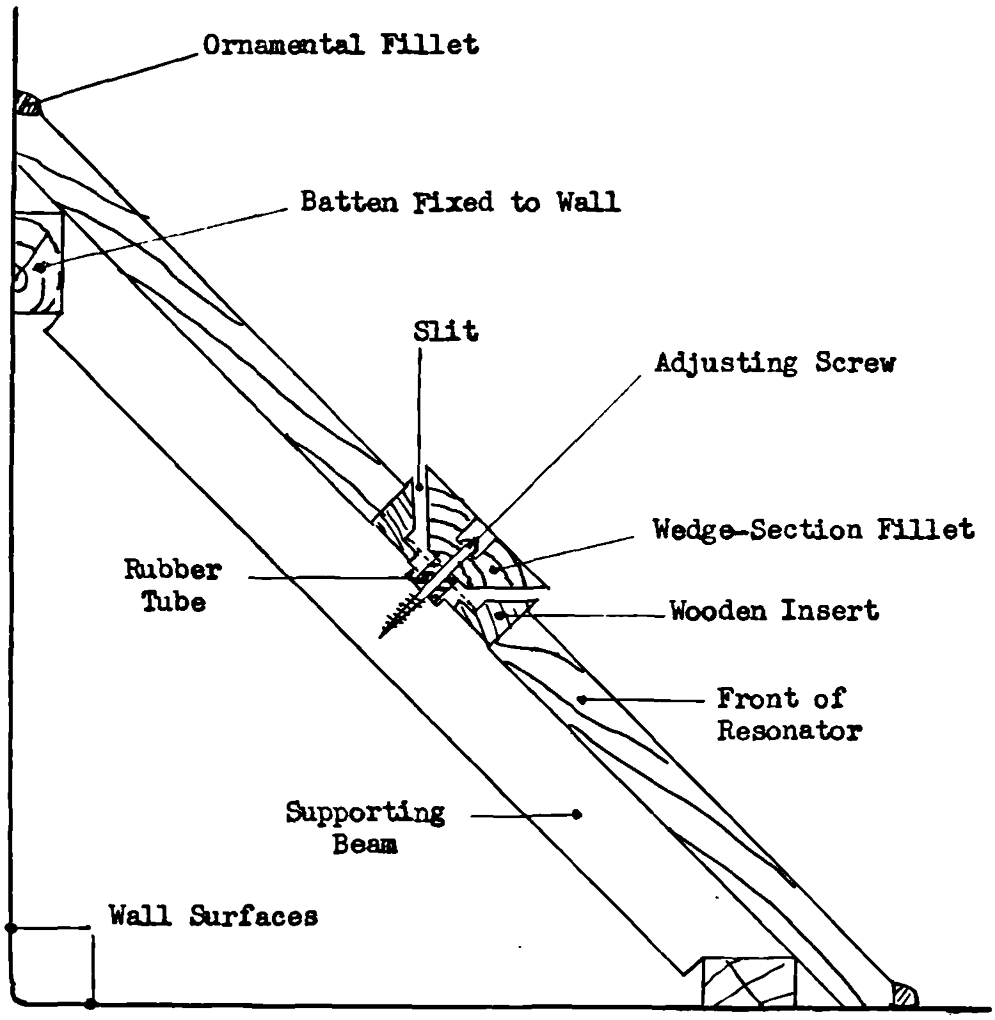


Fig. 6.3 Slot-Resonator with Adjustable Neck Conductance for Mounting in Corner of Room.
(Section through Adjusting Screw and main supporting beam) Scale approx. $\frac{1}{4}$.

The adjusting Screw moves the Wedge-Section fillet against the pressure of the rubber tubular stop, thus opening or closing the two slits.

occupied by doors and windows. This framework is ended at a suitable distance from each plan corner to allow for the mounting of a large slot resonator from floor to ceiling, of sufficient volume for efficient absorption at low colouration frequencies.

It is expected that these will be used only if the studios, when initially completed, show noticeable colourations requiring selective absorption. It will be appreciated from the discussions above and in this paper generally that prediction of colouration frequencies is insufficiently reliable to depend entirely upon narrow-band selective absorption and it will be necessary to provide enough general bass absorption to achieve a low reverberation time and consequertly less frequent incidence of severe colouration.

In the next two sections, types of absorber more suitable for general than selective absorption will be considered.

6.3 Membrane Absorbers

The resonant membrane low-frequency absorber has been treated in a paper by the author (Gilford 1952-3) which is bound with this thesis. They have been in widespread use in the B.B.C. from that time and in this section their relationship to small room problems and to other resonant absorbers will be examined.

Briefly, a membrane absorber consists of a thin flexible membrane or sheet of an impervious flexible material closing the front of an airspace. Sound pressure on the membrane causes is to vibrate, the amplitude being determined by its mass and the stiffness of the air in the space behind, thus dissipating energy from the sound by internal damping in the membrane or air space. The airspace may be damped by a filling of rockwool or other fibrous material. Such a construction usually exhibits a single resonance at a frequency for which the acoustic capacitative reactance of the air inside is equal and opposite to the sum of the reactances of the membrane and the radiation impedance and at this frequency the amplitude of vibration, and consequently the absorption, reach a maximum value.

The membrane material which has been used with the greatest success is common and bituminous roofing felt, of mass 0.24 g/cm^2, which may be used as a single layer or as two or three sheets laid together if a greater mass is required. The resonance frequency may be designed to have any value within the range 50 to 300 c/s or even higher.

In absorbing sound preferentially at a resonance frequency determined by a mass and a stiffness, membrane absorbers have features in common both with the Helmholtz resonator absorber already described and with the wooden or hardboard panelling which has been used, albeit unconsciously, for low freouency sound absorption in concert and assembly halls for many centuries.

Wood panelling, however, exhibits many modes of vibration with different frequencies of resonance, and therefore does not lend itself to accurate design or ready adjustment for a particular purpose. On the other hand (as shown by the author in the paper quoted above), the flexibility of the roofing felt membranes causes all modes other than the simplest, in which the displacement is in phase all over the membrance, to be suppressed.

Compared with the Helmholtz resonator, the membrane absorber is essentially a wide-band absorber. This becomes clear from the data given in the paper, together with the relations between bandwidth and Q given in 6.1 above. The radiation resistance of a circular piston of radius r is given by Crandall (1926) as

$$R_A = \frac{c}{\pi r^2}\left\{1 - \frac{J_1(4\pi r/\lambda)}{2\pi r/\lambda}\right\}$$

where λ is the wavelength of the sound, and J_1 the Bessel function of the 1st kind, order 1.

Thus the radiation resistance of unit area is

$$R_{A,1} = c\left\{1 - \frac{J_1(4\pi r/\lambda)}{2\pi r/\lambda}\right\}$$

Fig. 6.4 (a) shows the function plotted against the frequency for a piston of area 0.6 m^2. Curve (b) shows the bandwidth calculated on the assumption that an absorber of this size is perfectly matched to the radiation resistance, while curves (c) and (d) respectively show the bandwidth to be expected from single and double-ply roofing felt membranes using the constants for this material derived in the paper. The intersection of (b) with (c) and (d) shows the frequencies at which the single and double membranes are correctly matched.

It will be seen from the curves that, even with internal resistance too low for correct matching, the bandwidth of absorption of an absorber with an area of approximately 0.6 m^2

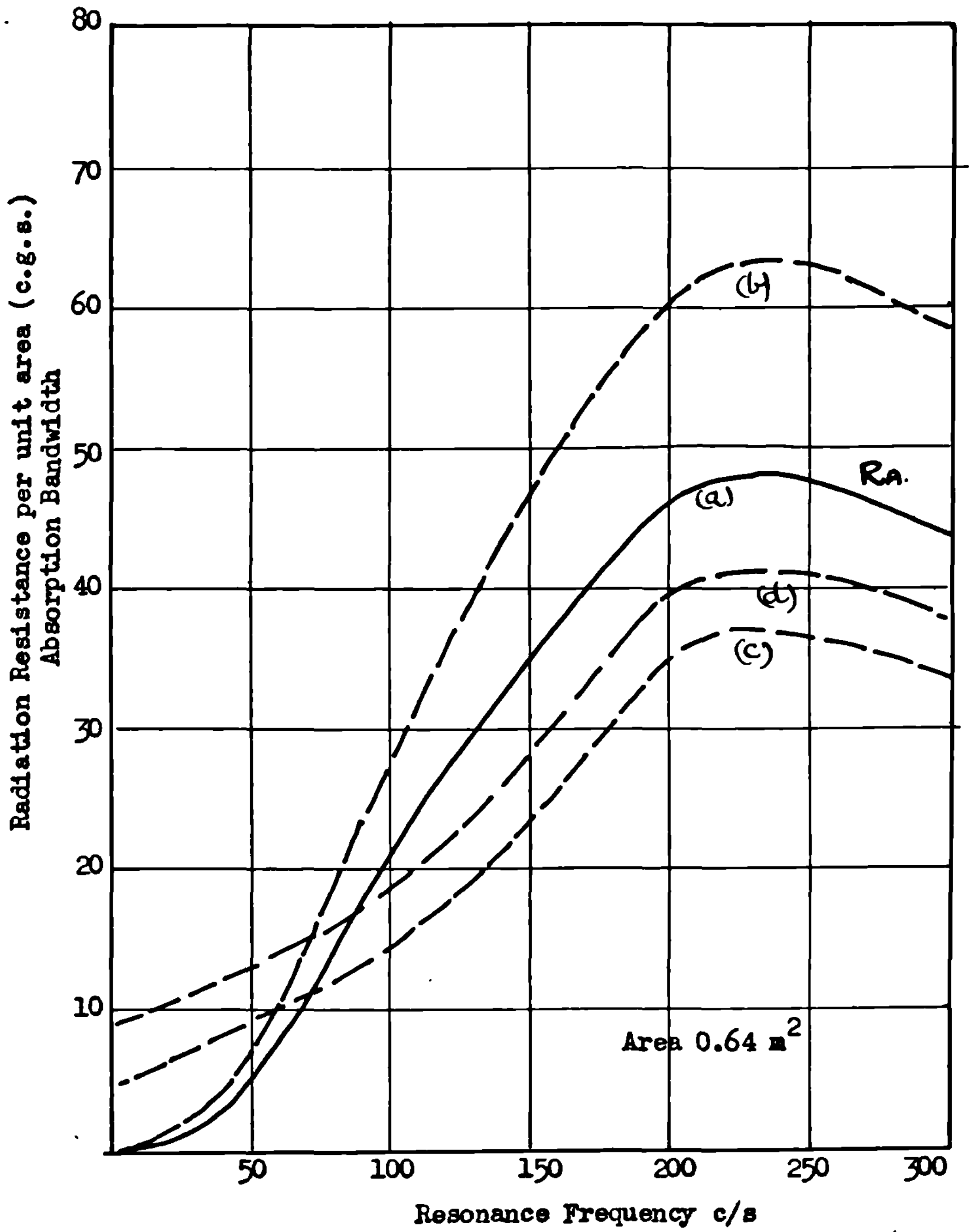


Fig. 6.4 Radiation Resistance and Bandwidth of typical Roofing-Felt Membrane Absorber.

(a) Radiation Resistance R
(b) Bandwidth assuming Correct Matching
(c) " of Single-Membrane Unit
(d) " " Double-Membrane Unit

is very much greater than that of a typical room mode. From a frequency of 175 c/s upwards the radiation resistance per unit is substantially the same as the characteristic resistance of a plane wave field. Therefore for absorbers of practical size the resistance cannot be reduced by a large ratio without reducing also the peak absorption. Therefore the only method of reducing the bandwidth would be to increase the mass of the membrane, but without increasing the stiffness or internal damping. A suitable material would be polyvinylchloride with lead loading but a search carried out in another connection failed to find a p.v.c. mix having suitable rheological properties.

If the absorber is made very small in comparison with the wavelength, as at low frequencies in Fig. 6.4, the radiation resistance per unit area will again be small, but to achieve this, it would be necessary to separate the individual absorbers to a considerable distance. Using single roofing felt, for example, it may be shown from the data of Fig. 6.4 and the methods outlined above that the radius of a circular absorber to give absorption at 150 c/s with a bandwidth of 10 c/s would be approximately 20 cm, the area 0.125 m^2 and the absorption equivalent to an area of 0.3 m^2 of perfect absorber. If 'matched' by introducing resistive material into the space the latter figure will rise to a maximum of 0.7 m^2. This is clearly a much less effective way of providing narrow-band absorption than the Helmholtz resonator treated above for which a figure of 78 ft^2 (7.4 m^2) was derived.

(A variant of the roofing felt membrane absorber introduced by A.N. Burd and the author (Burd and Gilford 1958) is the so-called bonded membrane absorber in which the membrane consists of a sheet of roofing felt stuck to a piece of 1/8" (3 mm) hardboard. The type is mentioned because it is now in widespread use. It is intermediate in properties between a true flexible membrane and a panel. Its bandwidth is similar to that of a roofing-felt membrane absorber and it is therefore suitable only for general low frequency absorption.)

6.4 Porous Absorbers

Absorbers consisting primarily of porous or fibrous materials through which the air particles are driven by sound pressure, losing their energy by viscous losses, are in general use for the reduction of noise and their general properties are well known.

At low frequencies the absorption coefficient of a porous layer of depth small compared with the wavelength, backed by a hard wall, is low because the particle velocity near the wall is low. In a diffuse random incidence field, the absorption coefficient of a thin layer of porous material reaches a first maximum, which may approach unity, at a frequency for which the wavelength is about eight times the distance of the layer from the wall. For example, at 150 c/s the layer should be 0.95 ft (29 cm) from the wall to give the highest absorption coefficient. After reaching the frequency of maximum absorption the coefficient tends to oscillate. The maximum is a very broad one but a narrower bandwidth is obtained by covering the absorber with a sheet of perforated material, particularly one of low-percentage perforated area. Fig. 6.5 shows two examples of absorption curves using perforated covers of 0.5% open perforation area over rockwool in two different depths. Perforated-front absorbers of this type have been treated by various authors, the fullest account having been published by Ingård and Bolt (1951). With low-resistance porous filling they show calculated bandwidths proportional to the frequency of peak absorption similar to those of roofing felt membranes. The peak absorption in their curves is higher for low-resistance fillings than for high, and vary from 0.6 to 0.95. The denser fillings give very large bandwidths.

These absorbers are therefore unsuitable for selective absorption of room modes. Their bandwidths are normally too great for adjustment of low frequency reverberation and it is necessary to use inconveniently great depths to achieve high coefficient of absorption within the colouration frequency range. Ingård and Bolt show that by partitioning the airspace behind the porous material so as to restrict flow to a direction normal to the surface it is possible to increase the coefficient at low frequencies. Nevertheless porous absorbers, with or without perforated covers, have lower

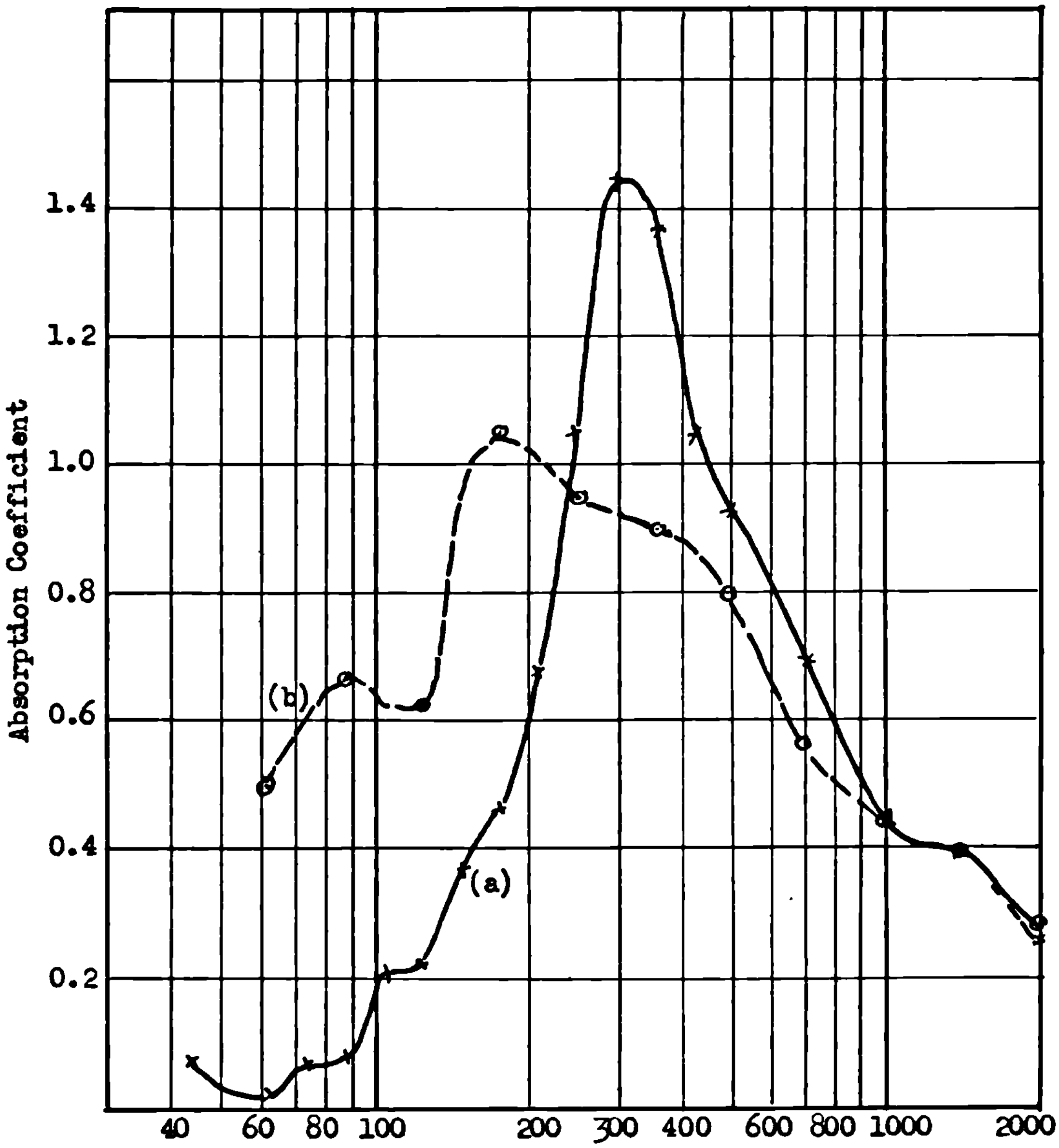


Fig. 6.5 Examples of Curves of Coefficient of Absorption obtainable with rockwool over a perforated cover.

(a) 1" Light Density Rockwool covered with 0.5% open area perforated Hardboard (7/64" to dia. holes in square array of side 1 5/16")

(b) 1" Dense Rockwool over 7" air space with 0.5% perforated Hardboard.

Samples 6 ft x 4 ft (1.83 m x 1.22 m)

absorption coefficients than membrane or Helmholtz absorbers at the frequencies with which we are here concerned.

6.5 Functional Absorbers

6.5.1 Introduction

From the consideration of low-frequency absorbers of the Helmholtz and membrane classes, it is clear that high total absorptions can be obtained from small objects by reason of diffraction. For this reason, remarkable claims have been made from time to time for absorbers consisting of three-dimensional shapes made from porous materials, hung freely in a room as opposed to mounted against the walls or ceiling. These have been termed "Functional absorbers" by Olson (Olson 1946) because, he argued, they are purely functional as sound absorbers, not requiring any other properties normally required by wall finishes. Olson gives results showing that the absorption of a compressed fibrous material was approximately twice as much when made up into double-conical shells and hung from the ceiling as when laid on the floor of the room as a single sample 72 ft^2 (6.8 m^2) in area.

There is no suggestion that these absorbers would have selective properties but since it was possible that they could be made highly efficient at low frequencies by suitable exploitation of the diffraction effects, they were investigated experimentally and theoretically as absorbers for the general reduction of reverberation time in the colourations region.

Some years ago, absorbers consisting of cubical shells of slotted hardboard lined with rockwool were used as wideband absorbers in a television control room at Alexandra Palace; the absorption coefficient of the material, measured by the reverberation method described below in Chapter 7, was less, except at very low frequencies, than that of a similar material laid flat on a hard surface. Figs. 6.6 (a) and (b) show the results of these measurements.

Abramchek and Maletskii (1959) using similarly shaped absorbers with perforated metal surfaces obtained even lower figures for the absorption per unit surface area of their absorbers (Fig. 6.7).

The author therefore carried out an approximate theoretical treatment of spherical and cylindrical shapes to determine whether any great advantage was to be expected from the free-hanging situation. This treatment follows below, together with some experimental results for cylindrical types which confirm the validity of the theory. The experimental work yields one result, however, which was not predicted; when mounted in the corners of a room the cylindrical absorbers give a high peak of absorption at a low frequency. This phenomenon and its possible application are discussed below, though there has been no opportunity yet to extend the experimental work or explain it satisfactorily.

6.5.2 Simple Theory of Functional Absorbers

It is possible to make approximate calculations to determine the maximum possible absorption to be expected from functional absorbers, and to compare their efficiency in the use of absorbing material with that of conventional wall-mounted absorbers.

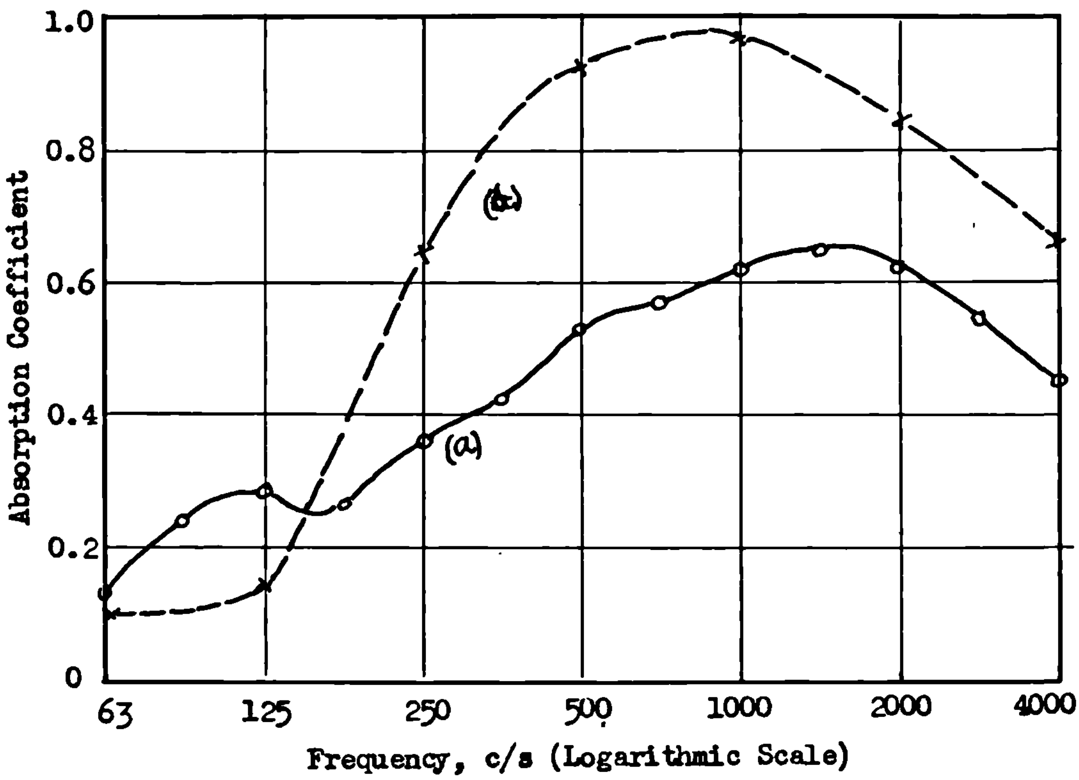


Fig. 6.6 Absorption Coefficient of Material for Cubical Functional Absorbers

(a) Used in functional Absorbers
(b) Measured as flat sample

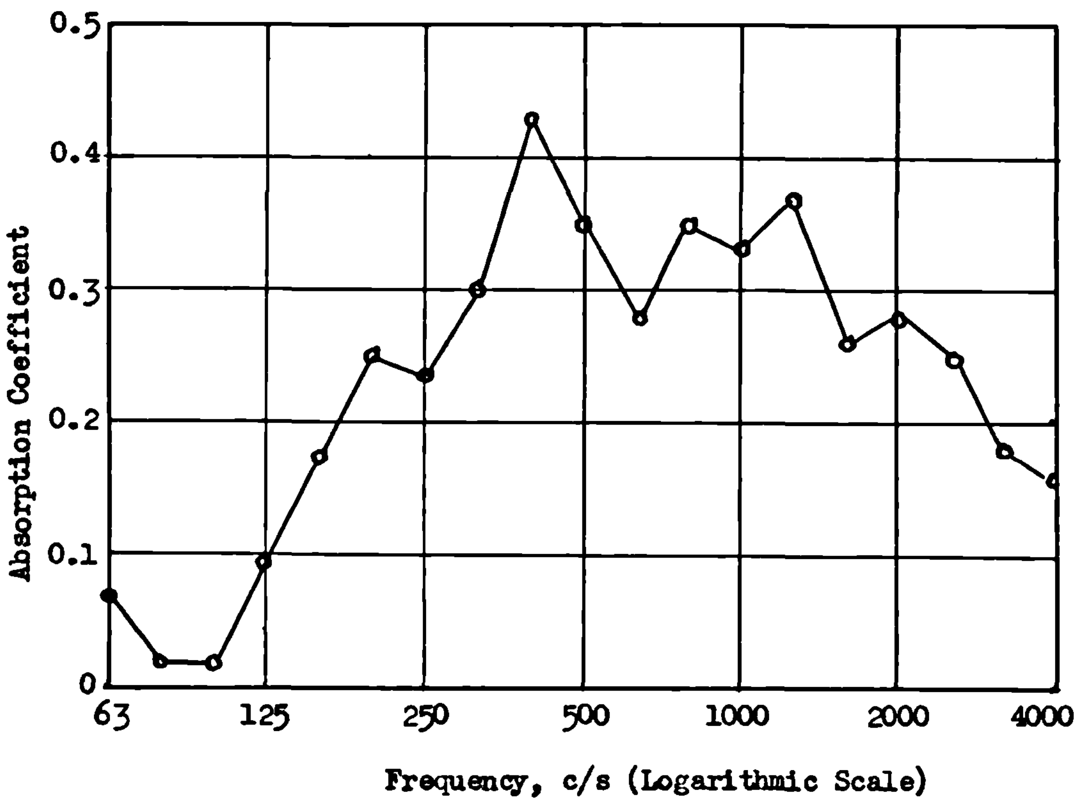


Fig. 6.7 Absorption Coefficient of Material used by Abramchik and Maletskii

(Measurements at one-third octaves)

Without trying to make exact calculations, one can derive this information by applying the general principles governing the absorption by a body of finite size, as discussed in connection with other types above.

Consider the behaviour of the material in one of these absorbers. We are interested primarily in very low frequencies where the wavelength of sound is large compared with the dimensions of the absorber so that the pressure of the sound, given reasonably diffuse conditions is almost uniform in amplitude and phase over all points of the surface. For completeness, we will also briefly consider high frequencies where the wavelength is small and the effects of diffraction may be neglected.

Two forms will be considered, one such as (a) or (b) in Fig. 6.8, and which may be represented for analytical simplicity by spheres, and the other a cylindrical or prismatic shape, Fig. 6.8 (c) which will be regarded as a long cylinder.

The calculations will be based on equivalent electrical circuits after the methods given in Section 6.1 above.

6.5.3 Spherical Functional Absorbers

We consider an absorber consisting of a spherical shell of a porous material, radius r. At low frequencies the radiation resistance is half that of a Helmholtz resonator or similar body buried in a wall because, instead of receiving sound from half-space only, it receives it from both hemispheres. The radiation resistance for the hole in the wall is given in Section 6.1 above as $2\pi\rho c/\lambda^2$ where λ is the wavelength of the sound, and therefore that for the

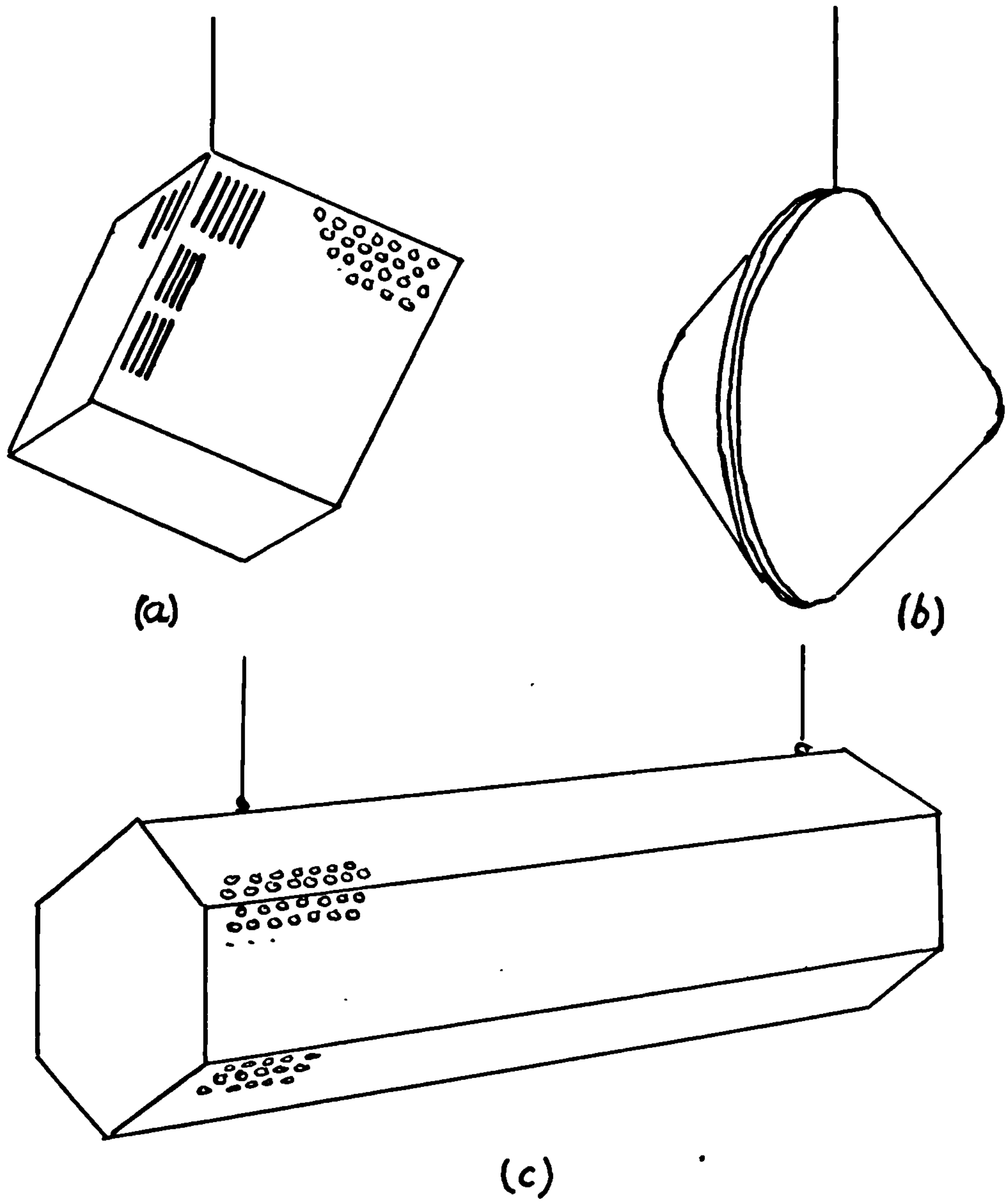


Fig. 6.8 Types of Functional Absorber

(a) Cubical (Perforated or slotted covers)
(b) Double Cones, Olson 1946 (Fibre)
(c) Prismatic (Perforated Metal)

functional absorber is given by

$$R_r = \pi\rho c/\lambda^2$$

Hence the internal resistance R_a per unit area of the absorber for a correct match is given by the equation

$$R_a/4\pi r^2 = \pi\rho c/\lambda^2$$

Whence, $R_a = 4\pi^2\rho c r^2/\lambda^2$

From Equation 6.4 the maximum possible absorbing cross-section for the absorber is given by

$$\rho c/R_r = \rho c\lambda^2/\pi\rho c = \lambda^2/\pi$$

Therefore, since the area of the surface of the sphere is $4\pi r^2$, the maximum effective coefficient of absorption is given by

$$\alpha_{eff} = \lambda^2/4\pi r^2$$

Suppose, for example the frequency is 200 c/s, $c = 3.4 \times 10^4$ cm/sec, $r = 20$ cm, $p = 1.23 \times 10^{-3}$ gm/cm^3, we have,

$$R_a = 23 \text{ dyne/cm}^2\text{/sec.}$$

$$\alpha_{eff} = 1.82.$$

In the above calculation we have neglected any effects of reactance. The figure of 1.82 would be valid only if the reactance components of the impedance were cancelled as in resonance condition. This could be achieved over a limited frequency range by the use of a covering of a perforated sheet material, the holes having inertance to cancel the capacitance of the volume inside the sphere.

Generally, however, there will be a series capacitive component, X, since the volume of the shell is finite, and this will reduce the flow of air through the shell and hence its absorption.

The volume of the sphere per unit surface area is:

$$\frac{4}{3}\pi r^3 \,/\, 4\pi r^2 = r/3.$$

The acoustic capacitance of this volume can be derived as follows. The compression modulus of a volume V_a for adiabatic volume changes is $\gamma P_o/V_a$ where P_o is the static pressure and γ the ratio of the specific heats of air (e.g. Richardson 1929, p. 224).

The acoustic capacitance is the inverse of this, and thus has a value of $r/3\gamma P_o$ for the volume of r/3. But $c^2 = \gamma P_o/\rho$ and hence the required acoustic compliance per unit area is

$$r/3c^2\rho$$

At 200 c/s this gives a reactance of 170 c.g.s. units per unit area in the case considered above.

From equations (6.2) and (6.3), the absorbing cross-section of an absorber having a series reactive component X is seen to be

$$\frac{P^2 R_f}{(R_r + R_f)^2 + X^2} \div \frac{P^2}{4\rho c} = \frac{4\, R_f \rho c}{(R_r + R_f)^2 + X^2}$$

By differentiating this with respect to R_f, we find that the maximum absorption occurs if

$$R_f = \sqrt{R_r^2 + X^2}$$

Putting

$$R_r = 4.5 \times 10^{-3}$$

$$X = 170/4\pi r^2 = 3.18 \times 10^{-2}$$

Then, $$R_f = 3.5 \times 10^{-2} = 175 \text{ c.g.s. per unit area.}$$

Substituting these values in equation (6.2) the maximum absorbing cross-section is found to be 0.46 cm^2 per cm^2 of surface; i.e., the effective absorption coefficient of the porous material used in this way is 0.46. This figure is comparable with that obtained from flat wall-mounted materials, as will be seen from the experimental evidence presented later.

At frequencies so high that diffraction may be neglected, the area of capture of a sphere of radius r is πr^2, irrespective of the direction of the incident sound. A disc of the same radius mounted on the wall, however, presents an area of $\pi r^2 . \cos\theta$ to sound incident at angle θ with the normal to its plane.

Imagine a hemisphere of large radius R_o in the sound field, assumed diffuse, with its centre at the centre of the disc. The sound incident at angle θ is then proportional to the area of an elementary ring of the sphere of infinitesimal width $R_o\, d\theta$ subtending an angle θ with the normal.

The sketch on the following page illustrates this hemisphere. The sound field is on the right and sound is incident equally from every element of solid angle surrounding the hemisphere, (Fig. 6.9).

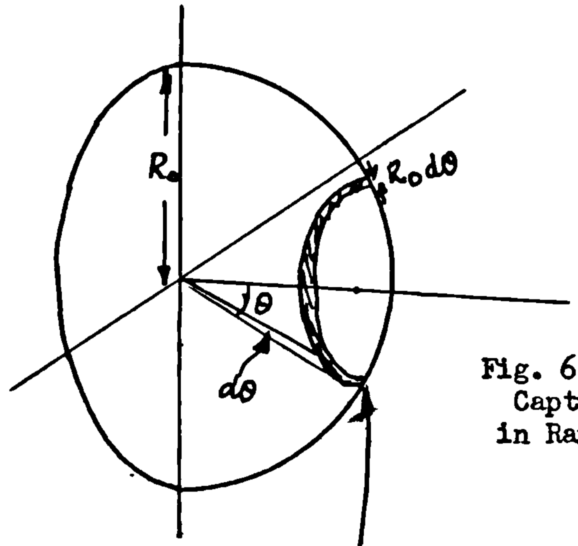


Fig. 6.9 Calculation of Capture Area of Disc in Random Sound Field

This area is $2\pi R_o^2 \sin\theta\, d\theta$

The mean area of capture for randomly incident sound may be shown then to be:

$$\frac{\pi r^2 \int_0^{\pi/2} 2\pi Ro^2 . \sin\theta . \cos\theta \, d\theta}{\int_0^{\pi/2} 2\pi \, Ro^2 \sin\theta \, d\theta}$$

$= \frac{1}{2}\pi r^2$

i.e. one half the actual area of the disc.

This mean area applies only to sound coming from one hemisphere, whereas the sphere hung in free space receives sound from both hemispheres. The effective areas of capture for the sphere and the disc are thus in the ratio of four to one, i.e. in the same ratio as their actual surface areas.

A given area of material will therefore receive and absorb sound energy at the same rate whether it is mounted flat on a wall or formed into a spherical functional absorber.

6.5.4 Cylindrical Functional Absorbers

We will apply the same methods to a long but finite cylinder as being an approximation to the prismatic or cylindrical types such as that shown in Fig. 6.8 (c). At low frequencies the absorber may be regarded as a long narrow strip. The radiation resistance of unit length of a long strip in a plane is $\pi\rho c/\lambda$ (Bruel, 1951) and hence for a line source in free space it is half this;

i.e., $R_r = \pi\rho c/2\lambda$

The maximum absorption with a matched resistance is therefore,

$$\rho c . 2\lambda/\pi\rho c = 2\lambda/\pi$$ per unit length, and the maximum absorption per unit area of surface (effective absorption coefficient of the surface) is

$$\frac{2\lambda}{\pi} \cdot \frac{1}{2\pi r} = \frac{\lambda}{r\pi^2}$$

Proceeding as for the sphere in the previous section, we have for f = 200 c/s and r = 20 cm., taking into account the reactive component due to a volume per unit surface area r/2, we find $\alpha_{eff} = 0.56$ and $R_f = 114$ (optimum).

Thus the cylindrical shape gives a higher efficiency in the use of material than does the sphere and it requires a less dense material for optimum match. (R_f = 114 compared with 175 c.g.s. units.)

At high frequencies, the mean capture area of a long cylinder of length l may be shown by a similar argument to that used above for the sphere to be:

$$\frac{1}{\pi R_o^2 \delta_o} \int_0^{\pi/2} 2r\ell\pi R^2_o \cdot \cos^2 \theta . d\theta = \tfrac{1}{2}\pi r l, \text{ which is one}$$

quarter of its actual surface area. As the cylinder we are considering is in free space its effective mean capture area compared with a wall-mounted shape is doubled and becomes in effect half its surface area, which is identical with the mean capture area of a wall-mounted strip of the same surface area.

To summarise, a cylindrical functional absorber may be expected to have at best an absorbing cross-section at high frequencies equal to a strip of wall-mounted material of the same length and area. At low frequencies for which the wavelength is much greater than the radius of the cylinder it will have a better performance, area for area, than a sphere.

Very similar conclusions apply also to medium frequencies. Calculations by the author for this case will appear elsewhere, but as they depend on other evidence and considerations, requiring exposition at some length, and as their relevance to the present discussion is questionable, they will not be given here.

6.5.5 Measurements on Functional Absorbers

The calculations given above are here compared with measurements made previously on prismatic functional absorbers constructed by the Darlington Insulation Company. These were, for convenience in manufacture, made hexagonal in cross-section, the outer shell being of perforated aluminium of which the perforations amounted to 25% of the whole surface area. Their length was 6 ft (18 m) and the side of the cross-section was 8¼ in. (21 cm). The shell was lined with rockwool 1½ in. (3.7 cm) thick with a flow-resistance of 40 c.g.s. units. Various modifications were made to obtain the best performance over the whole audio-frequency range, and in the course of the experiments membranes of paper and calico were interposed between the perforated shell and the rockwool to provide inertance with the object of increasing the absorption at low frequencies. Fig. 6.10(a) shows the results with the rockwool lining only while curve (b) shows the best obtained with added inertance layers. Curve (c) is for one inch (2.5 cm) of rockwool (Rocksil H.D.S.) over 4-inch (10 cm) airspace with 25% perforated cover. It will be seen that it is only at low frequencies that the functional absorber shows any advantage over the flat sheet. At high frequencies, as predicted in section 6.4.4 above the results are very similar, with the advantage to the wall-mounted absorber.

The results published by Olson (1946), to which reference was made above, do not conflict with those of the present investigation. It will be appreciated that Olson's comparison was between a single flat sheet of material 6.8 m^2 in area (actually made up of smaller sheets placed edge to edge and nailed to battens) and a large number of small pieces of the material separated from each other. But, as shown earlier in this chapter, very much higher effective absorption coefficients may be obtained from wall-mounted absorbers of low internal resistance if they are subdivided into small areas. Kuhl's results (Kuhl 1960) for a typical porous material which he

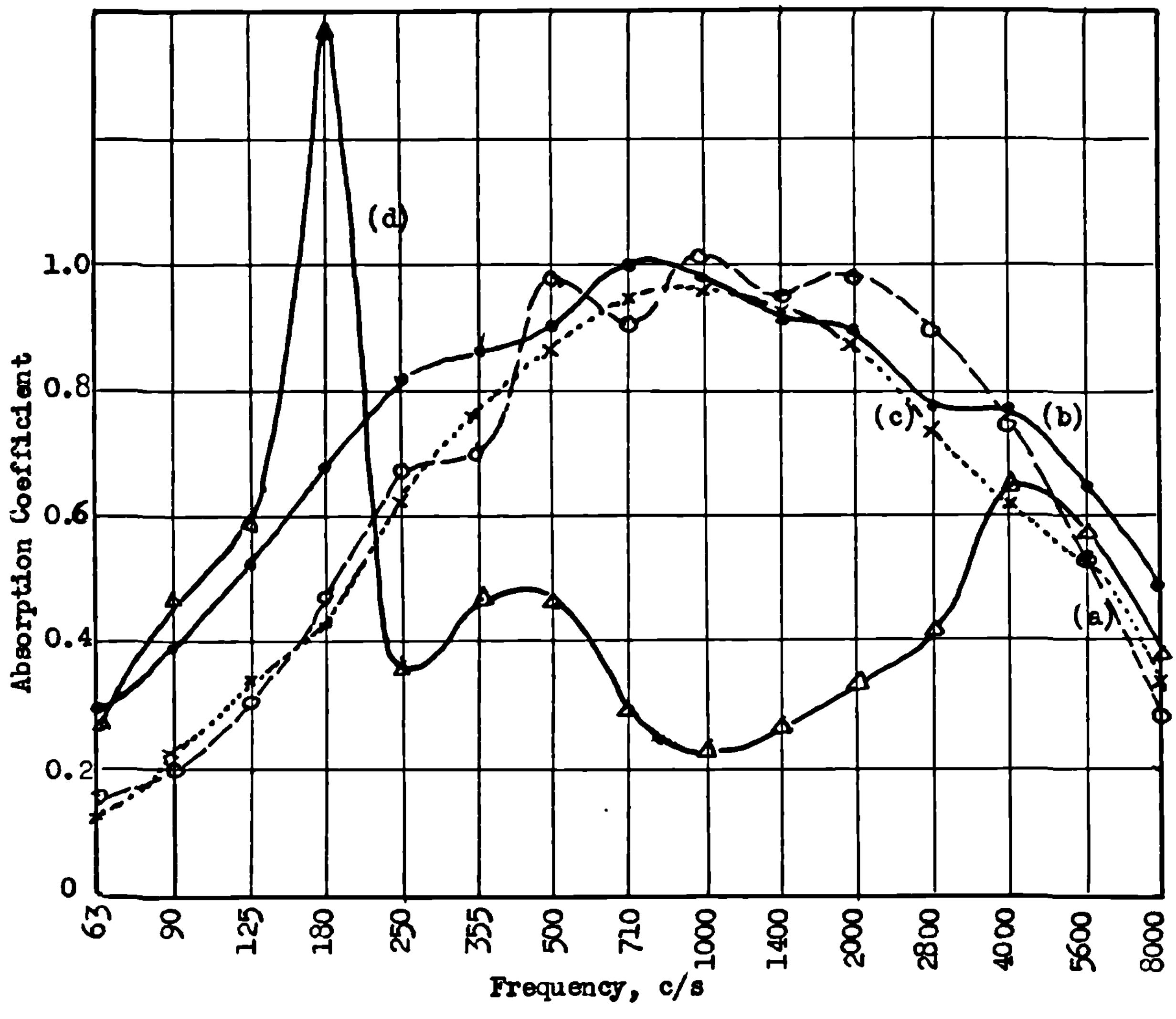


Fig. 6.10 Measured Effective Absorption Coefficient of Material of Prismatic Functional Absorbers

(a) Rockwool behind Perforated Metal
(b) As (a) but Calico Inertance Layer added
(c) 1 inch (2.5 cm) Rockwool over 4 inch (10 cm) Airspace
(d) As (b) but tested in corners of room

subdivided by stages into 20 small pieces are sufficient to account for the whole difference found by Olson between functional absorbers and a whole area of the same material. With the sample subdivided into pieces 1 m square, he obtained a maximum absorption coefficient of 1.53. Olson's largest cones had a surface area of 0.46 m^2 each, and the smallest, 0.11 m^2.

Fig. 6.10 (d) shows the effective absorption coefficient when the absorbers are tested in the extreme corners of the room. The peak at 180 c/s suggests that corner mounting has produced a resonance within the airspace behind the absorber but the means by which this was effected are not clear. There will be a capacitative component of the impedance, equivalent to the compliance of the air in the corner enclosed by the absorber, and the perforated covers to the rockwool would undoubtedly have an inertance; but rough calculations on the magnitude of these components indicate a much higher resonance frequency, around 700 c/s. This phenomenon would repay further study, in particular to find the bandwidth of the absorption and to explain the mechanism. It is rather doubtful however whether it is of any particular use in the present form for the suppression of colourations; it shows no features of practical importance not shared by Helmholtz or membrane absorbers, and the construction is more expensive than either. There remains the possibility however that further investigation would reveal advantages not at present suspected.

With this possible exception the theory and measurements of this section do not encourage the belief that furctional absorbers are particularly useful either for suppression of colourations or for general low frequency absorption, though they would find application where wall or ceiling space was not available.

6.6 General Discussion of Sound Absorbers

In this chapter an attempt has been made to examine all the most efficient types of sound absorber with respect to their use for the suppression of prominent modes. This may be achieved by two distinct methods.

The first is by the use of an absorbent having a bandwidth similar to that of the undamped mode and exactly tuned to the modal frequency. By suppressing a single mode without affecting the rest of the spectrum, however, neighbouring modes will become more prominent and will require suppression in the same manner. As initial treatment of a talks studio it would be necessary to provide resonators tuned to a great number of possible colouration frequencies and in general this would be impracticable.

As a remedial measure, however, selective absorption of this kind has possibilities which have not yet been fully exploited. To make efficient use of these possibilities, however, additional study is needed to find methods of determining accurately the frequencies of the primary and subsidiary colourations and the narrow-band spectrometer described in Chapter 5 is being developed to this end. More experiment is required in the construction of resonators of the slit type since, although they lend themselves to simple adjustment, they tend to give low values of Q.

The Helmholtz resonator is the only absorber in use to-day which is capable of yielding high enough values of Q for true selective absorption; modified forms of membrane absorber to give higher Q values are, as shown earlier in this section, theoretically possible, but have not yet been tried.

The second method is by the use of resonant absorbers to damp the whole of the colouration frequency range, the bandwidth being considerably greater than that of a mode but small enough to enable a uniform reverberation time to be attained by the use of absorbers with differing frequencies of maximum absorption. For this purpose, as shown in the analysis above, membrane absorbers of moderate size (about 0.5 to 1 m^2) are particularly suitable, as would be plane arrays of Helmholtz resonators of a similar size, suitably damped.

Porous absorbers are not particularly suited to low frequency absorption in small studios because rather large depths are required for efficient absorption, and bandwidths are generally too large at low frequencies.

The remaining type, which has been investigated above in some detail, is the functional absorber. This seems to have no special application for low frequency absorption. A purely experimental discovery that when mounted on a corner a functional absorber behaves as a resonant system is worth further study; since, however, the corner space in any room is strictly limited it does not seem likely that this discovery will lead to any serious practical applications.

CHAPTER 7

THE INFLUENCE OF POSITION ON THE EFFICIENCY OF ABSORBERS

7.1 Introduction

In the previous section it was concluded that, in designing a small room for acoustics free of colourations the use of wide band absorbers to keep the reverberation time short at low frequencies was more practical than attempting to predict colouration frequencies and absorb sound selectively at those frequencies. Therefore, a resonant absorber with a bandwidth longer than the average spacing of axial modes should be used. Having regard to the fact that only a limited area of the room surfaces can be used for the installation of bass absorbers, the floor, doors, windows, skirtings and points for installation of technical services not being available, absorbers having high coefficients are often necessary.

Moreover the construction of membrane absorbers or, for that matter, any low-frequency sound absorber, is expensive and for this reason it is desirable to find out how they can be used most efficiently. The influence of position on the performance of single Helmholtz resonators for selective absorption of individual modes was briefly touched upon in Section 6.1 and it was decided to find out whether the position in the room had a similar effect on absorption by broad-band absorbers operating on several adjacent modes.

Every mode necessarily creates maximum pressure along one or more pairs of surfaces, in at least four edges and in all eight corners of a rectangular room. It is therefore to be expected that corner- or edge- mounted absorbers of any type will be most effective, provided that any necessary modifications are made to ensure correct matching of the absorber to the radiation impedance at the chosen position.

Theoretical work on the statistical variations of sound pressure with position in a room has been carried out by Waterhouse (1955) who derives the sum of the mean squared pressures of a large number of interference patterns caused by wave trains incident at all angles. He shows that the pressure against a wallis greater by a factor of $\sqrt{2}$ (3dB) than the mean pressure measured in the centre of the room; at an edge formed by the junction of two surfaces the pressure is doubled (6dB increase) while in a corner of the room the increase is 9dB.

The corresponding mean potential energy densities are twice, four times and eight times the values in the centre of the room, since energy density $\propto$ pressure^2.

Whether we are considering pressure or energy density, he shows that the region where the value exceeds that at the centre of the room extends to a distance r given by

$$2kr \sim 4 \quad \text{where } k = 2 / \text{wavelength } (\lambda)$$

i.e. $\quad r \sim \lambda/\pi$

Within this distance of the edges or corners an absorber will experience a sound pressure greater than that in a typical position on a wall, attaining maximum values of $\sqrt{2}$ and 2 times that at the wall face when actually at the edge or corner.

Waterhouse quotes experimental confirmation of these results from experiments by London (1941) who had measured mean squared pressures near to corners and edges. Much more detailed confirmation was given by Wöhle (1956) who gives the following figures.

TABLE 7.1

Increase of Mean Pressure in Room at Walls, edges and corners (Wohle 1956)

	37.5 - 75 c/s	200-400 c/s
At Walls (compared with centre region)	3.2 dB	2.7 dB
At Edges	6.2 dB	6.0 dB
In Corners	9.4 dB	8.4 dB

Waterhouse shows that the results are substantially unchanged if, instead of pure tone of wavelength λ , a band of noise up to an octave in width, but with the same mean wavelength is used. The pressure in the extreme edge or corner is unchanged and the variation of pressure with distance from the corner follows a similar form. Therefore the results may be taken to apply to bands of noise, or to frequency-modulated tone such as is often used for acoustic measurements in rooms. They apply likewise to resonant absorbers of small or moderate absorbing bandwidth.

Finally, he calculates the total extra energy contained in the edge and corner patterns as a fraction of the product of the room volume and the mean energy density in the central region of the room where the interference patterns are negligible.

For a numerical example in which $\lambda = 8$ ft (142 c/s) and the room is a cube of side 20 ft, he finds this ratio is 1.3 dB. Using these same data, the contributions of the interference patterns at the walls, the edges and the corners are in the ratio

$$1 : 0.127 : 0.042$$

These figures imply, in other words, that only about 0.127/2 or 6.35% of the wall surfaces may be regarded as edge positions and 0.042/4 or 1.05% as in the corners. The advantage to be gained by putting membranes or porous absorbers in such positions is therefore slight, the advantage would vanish entirely at high frequencies.

Nevertheless, further examination is warranted for Helmholtz absorbers of which the entries can be designed to be right in a corner and where the radiation impedance is an important characteristic of the design. These resonators will be considered in the next section below.

7.2 Helmholtz Resonator Absorbers

The effect of position and arrangement on the efficiency of Helmholtz Resonators has been dealt with by the author (Gilford 1952), Ward (1952) and Wöhle (1956, 1957, 1959). The most comprehensive work is by the last-named author, who treats single resonators and resonators arranged in infinite and finite line arrays.

Dealing with single resonators, Wöhle first calculates the statistical pressure increases on surfaces and in edges and corners, as in Section 7.1 above, and then derives expressions for the radiation resistance of a resonator in these positions.

He shows that if W is the radiation impedance of a small radiator such as the orifice of a Helmholtz resonator in a wall, the impedance at distance r from an abutting wall is

$$W_{edge} = W\left(1 + \frac{\sin \frac{2\pi rn}{\lambda}}{2\pi rn/\lambda}\right)$$

Right in the edge we have $r = 0$ and hence

$$W_{edge} = 2W \quad \text{Since } \sin 2\pi rn/\lambda \,//\, 2\pi rn/\lambda \longrightarrow 1.$$

Similarly, the introduction of a third wall at right angles to the first two gives a result

$$W_{corner} = 4W$$

It will be noticed immediately that the radiation impedance increases in an edge or a corner in the same ratio as the energy densities increase, and this indicates simply how the behaviour of a resonator is modified by its position. In Section 6.4.2 above it was shown that the maximum absorption of an absorber at resonance is given by

$\rho c/R_r$ where R_r is the radiation resistance.

If the sound pressure at the absorber is increased in the ratio h : 1 compared with the mean pressure at a wall surface, its maximum absorption will be $h^2\rho c/R_r = E\rho c/R_r$ where E is the ratio of the energy density at the chosen position to that at a plane surface.

Hence, since R_r increases at the edge and corner positions in the same ratio as the energy density, the matched maximum absorption at resonance is independent of the position. However, if R_r is increased, the internal resistance of the resonator must be increased in the same ratio to preserve the correct matching and therefore the bandwidth is increased in the same ratio.

Fig. 7.1 represents this result. The curves are the calculated absorption/frequency characteristics of a resonator in the three positions, its internal resistance having been matched in each position to the radiation resistance to achieve maximum absorption at resonance. The peak absorption occurs at a frequency below the resonance frequency, however, and attains somewhat higher values in the case of edge and corner mounted than with wall centred absorbers, i.e. there is an advantage both in bandwidths and peak absorptions.

For a line array of m holes, distance a apart, Wohle gives as the maximum possible absorption $\frac{ma\lambda}{\pi}$ or λ/π per unit length. In this instance, placing in a corner has no significance; placing at an edge gives a similar increase in bandwidth to that noted with a single hole resonator, without any increase in maximum possible absorption, but the bandwidth is already greater than that of a matched single-hole resonator. The same remarks apply to resonators in which the opening to the air is a long slit. (See Section 6.2).

7.3 Membrane and Panel Absorbers

The second class of resonant absorbers, i.e. membrane and

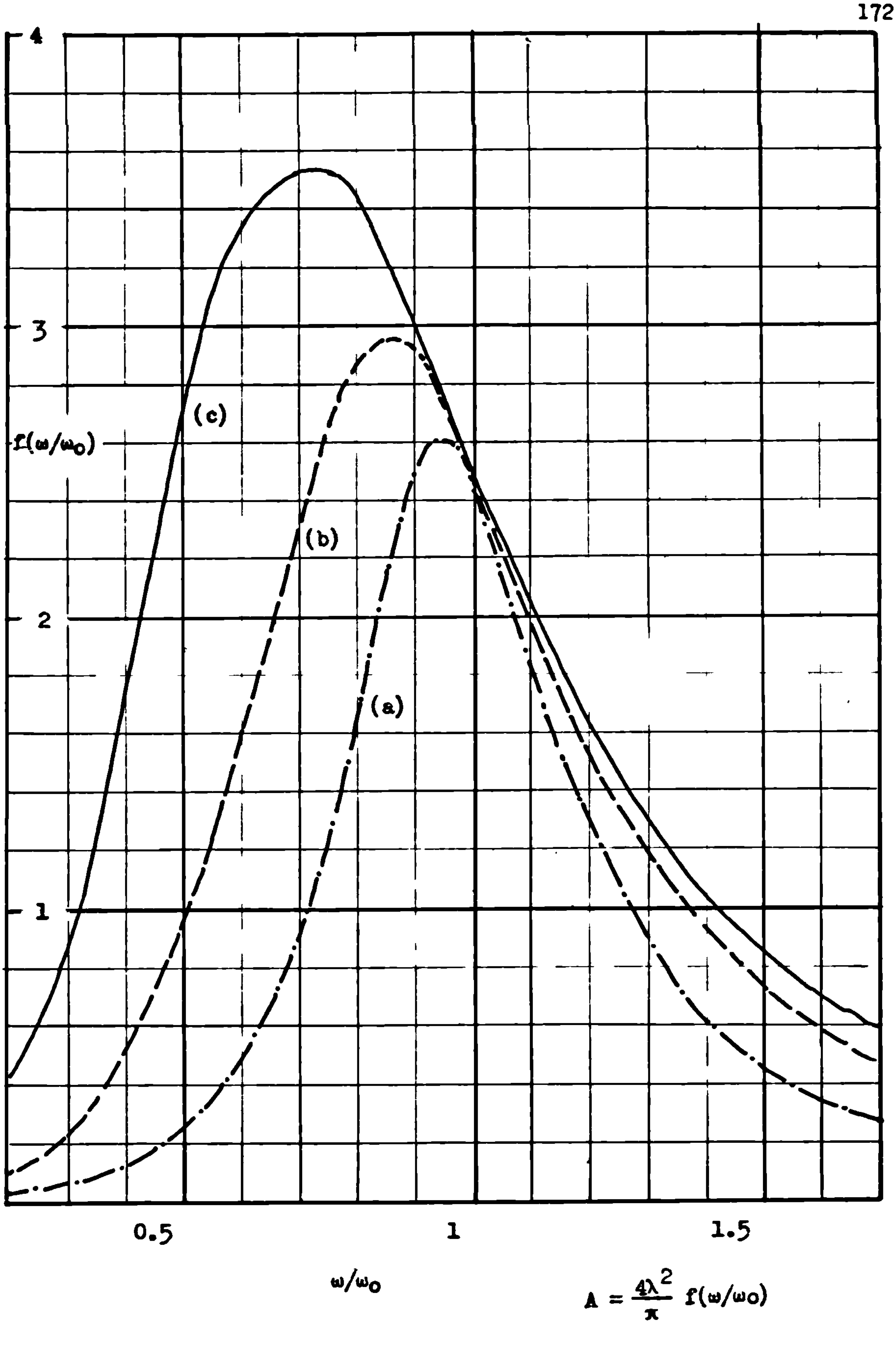


Fig.7.1 Absorption Characteristics of Helmholtz Resonator
(a) On Wall Face (b) In Edge (c) In Corner

panel absorbers in which the mass of the resonant system is a solid lamina, presents only one new feature; the radiation resistance alters with the size and this must be taken into consideration in deciding the amount of corner or edge space which can be used with advantage compared with the use of open wall areas.

The limiting case of a very small membrane in an edge or a corner would clearly approximate to that of a single Helmholtz resonator which was treated in the last section above, and we should therefore expect no increase in the maximum possible absorption, but an increase in the bandwidth compared with an open-wall situation.

It is therefore of interest to investigate the practical case of a square membrane unit of typical size placed on a wall with two of its sides along two of the three edges of the room meeting at a corner. In considering a numerical case, approximations will be made as follows:

For calculations of the mean pressure of the sound-field, the absorber will be treated as a quadrant of a circular lamina, with its centre at the corner of the room and area of 0.6 m^2. The radius will then be 88 cm.

Fig. 7.2 curve (a), shows the pressure at the circumference of the quadrant taking the mean pressure in the central region of the room as unity, re-drawn from the data in Waterhouse's paper and plotted against 2kr where $k = 2\pi/\lambda$ and r is the radius of the quadrant. Curve (b) shows the ratio of the mean sound pressure over the quadrant to that at the centre of the room derived by numerical integration of curve (a). The points on the 2kr scale corresponding

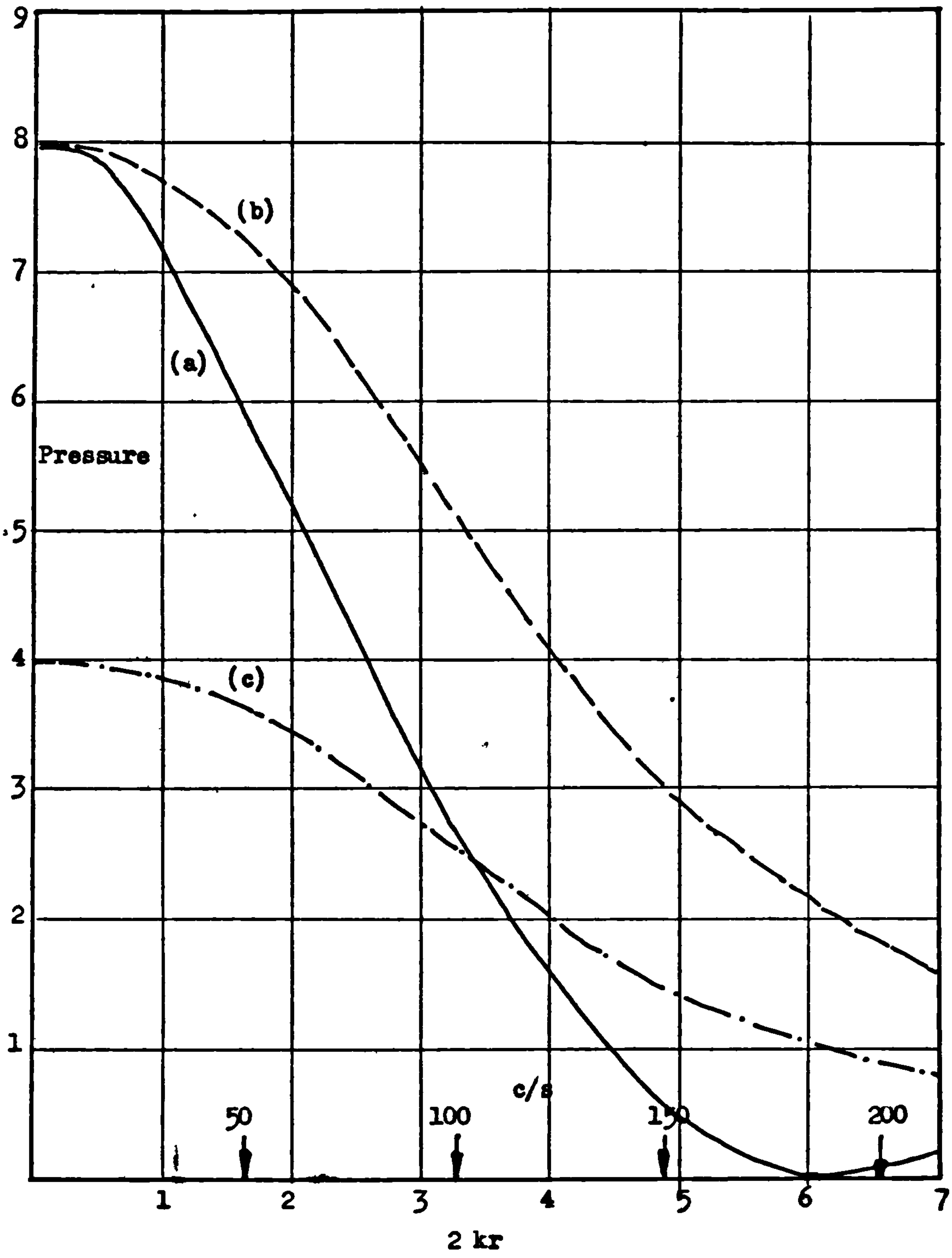


Fig. 7.2 Pressures on Quadrant Absorber in Corner of Room (Expressed as ratio to pressure in centre of room)

(a) Pressure at edge of Quadrant (From Waterhouses (1955))
(b) Mean pressure over Quadrant
(c) Ratio of Mean pressure to pressure on open wall

The scale above the 2 kr axis shows the corresponding frequencies. r = 88 cm to frequencies of 50, 100, 150 and 200 c/s are marked. Since the normal location of a membrane absorber is on a flat wall not particularly near to an edge or corner, the pressure on it is twice that in the central region of the room. Curve (c), in which the ordinates of (b) are divided by 2, therefore shows the ratio of the mean pressure on the quadrant to the pressure on a wall remote from a corner.

Now the quadrant will form two primary images in the adjacent walls and one secondary image. Together they form a circle of radius 88 cm. The radiation resistance of this disc at the frequencies marked on the figure and at 250 c/s are tabulated in Table 7.2 below, in column 2; they are derived from the formula of Crandall given in Section 6.3 above. Column 2 shows the resistance multiplied by 4 giving the value for the quadrant itself.

(1) c/s	(2) Resistance of 88 cm radius circle	(3) Resistance of original quadrant Rq	(4) Mean Pressure (Fig.7c)	(5) Mean Pressure / Rq	(6) Ra	(7) $\frac{1}{Ra}$
50	0.4	1.6	3.6	2.25	0.5	2.0
100	0.8	3.2	2.5	0.78	2.0	0.5
150	1.2	4.8	1.5	0.31	3.9	0.26
200	1.5	6.0	0.9	0.15	5.6	0.18
250	1.8	7.2	0.7	0.10	6.9	0.14

TABLE 7.2. COMPARISON OF MAXIMUM ABSORPTION BY TYPICAL MEMBRANE ABSORBER IN A CORNER AND ON A WALL AWAY FROM CORNER

Column 5 shows the result of dividing the mean pressure from Fig. 7 (c) by the radiation resistance, giving a number proportional to the maximum possible absorption. The last column (7) is the similar quotient for a circular wall-mounted absorber of radius 44 cm - i.e. the same area as the quadrant.

A comparison of columns (5) and (7) shows that, like the Helmholtz resonator absorber, the maximum possible absorption is improved at low frequencies by the corner mounting. The improvement is, however, not as great as the increase of pressure in the corner. There is an advantage in bandwidth also which will be seen from a comparison of the radiation resistance shown in columns (3) and (6). At low frequencies the increase in bandwidth is great but it diminishes steadily at higher frequencies so that above about 150 c/s the absorber will behave very similarly with regard to maximum possible absorption and bandwidth whether mounted in a corner or not.

7.4 Porous Absorbers

Deep porous absorbers would be expected to show very similar behaviour with location in the room to the membrane absorbers considered above. However they differ in having no pronounced resonances and are more difficult to threat theoretically. It was therefore decided to investigate them by purely experimental procedures and this work will be described in the next section. In general the results would be expected to be applicable also to membrane absorbers.

7.5 Experimental Work on the Effect of Position on Absorption

The experimental work on the effect of the position of an absorber in the room was carried out by the well-known reverberation method.

The room which was used is nearly cubical in shape, with dimensions 3.22 x 2.95 x 2.88 m and a volume of 27.4 m^3. It is entirely surfaced with glazed tiles which are cemented to a concrete floor and to 30 cm thick concrete walls and ceiling. The walls are restricted on the outside by sand forming a filling between them and outer brick walls, which serves to damp any resonant vibrations. The ceiling has a layer of sand 10 cm thick lying upon it for the same purpose. There is a heavy door with tapered seals and a steel inner surface centrally placed along one side of the room, and opposite to this a cavity into which a loudspeaker is sunk. The samples are held against the wall surface by wires attached to screws driven into the walls.

In the reverberation method of measurement of absorption, the reverberation time of the room is measured at a number of frequencies, first with the room empty and then with the samples mounted in place. The measurement of reverberation time is carried out using the apparatus and method described by Somerville and Gilford (1952). Since the reverberation process depends upon the excitation of several room-modes and the measurement of the pressure/time relationship represents the sum of the pressures in the various modes excited, the reverberation time must be measured at several different positions in the room. All of these will, in general, represent different degrees of excitation of the relevant room modes. Normally this is carried out by making measurements at the series of frequencies in one microphone position, then repeating it in a series of other positions. However, different loudspeaker positions will equally well give different excitation patterns for the room modes and to this extent, loudspeaker and microphone positions are reciprocal in effect. To increase the number of readings which can be made in a given time, therefore, a different method was devised. Three microphones were hung in permanent positions in the room. The positions were chosen in such a manner as to ensure that as many as possible of the simpler room modes were significantly excited and therefore detectable at each point. A second loud-speaker was also introduced in a similarly chosen position. The two loudspeakers and the three microphones were then connected to a rotary switch so that each of the six combinations was selected by one position of the switch. The reverberation time is read from the cathode ray oscilloscope for each of the switch positions by the method described in the paper referred to above and transferred to an electrical printing-adding machine. When the six results have been printed, they are added and the frequency is changed to the next value. This method reduced the time taken for a complete reverberation characteristic determination to about a quarter of the time taken by the normal method.

The total absorption of the Reverberation Room was calculated with and without the samples, using the formula (Eyring, 1930)

$$= \frac{0.049\ V}{-S\ \log\ (1-\bar{\alpha})}$$

where V = Volume of Room, S = total room surface area,

$\bar{\alpha}$ = mean absorption coefficient of all surfaces,

i.e. $S\bar{\alpha}$ = the required total absorption.

Eyring's formula has been found to be more accurate than that of Sabine (1922) - the absorption due to a sample was obtained as the difference between the pairs of values of total absorption.

This room has no deliberate perturbations of its surface to provide diffusion of the sound field which is necessary for measurements of absorption coefficient. Venzke (1956) has shown that sufficient diffusion is caused by the presence of samples only, the edges of which cause diffraction of the sound waves, provided that there are samples on at least two walls. This is the experience of the author and verification was afforded by Randall and Ward (1960). Nevertheless, the circumstances of the present experimental work made it appear doubtful whether there would be satisfactory diffusion.

Firstly, as will be seen below, the samples would necessarily be smaller than those desirable for usual measurements of absorption coefficient, and secondly it was not known whether equally good diffusion would result from the presence of the samples in each of the special positions with which we were concerned. To ensure adequate diffusion at all times, sheets of hardboard were therefore hung in the room after the manner proposed by Kosten (1959). The sheets were of sizes 8 ft x 4 ft (2.4 m x 1.2 m) to 4 ft x 4 ft (1.2 m^2) and were hung from the ceiling by wires which were adjusted to make the total projected area of all the sheets on each surface of the room roughly proportional to the area of the surface; (i.e. if there were only one sheet the direction cosines of a normal to its plane would be proportional to the areas of the room surfaces perpendicular to the three axes). The absorption of the hardboard was small in comparison with that of the room surfaces and negligible compared with that of the absorbing samples. Fig. 7.3 shows how the sheets were hung. They could be taken down when not required, and measurements were made for all positions of the absorbers, both with and without the hardboard sheets. Fig. 7.4 shows the measuring apparatus. The oscilloscope screen is at 8, the sequential switch at 3 and the adding machine at 10.

The effect of position on the performance of strongly resonant absorbers has been dealt with in the preceding sections 7.2 and 7.3. The present experiments were therefore confined to a porous absorbing material, on which no previous work has been reported.

The requirement that the samples were to be tested in the corners of the room was the main factor in determining their design. This requirement implies that all parts of the sample should be within λ/π, or, say approximately a quarter wavelength of the corner at the highest frequency in which we are interested, since it is only within this distance of a corner or edge of the room that the sound pressure is higher than the mean pressure in the rest of the room. At greater distances it rapidly tends towards the average value for positions remote from the walls. Further, if the results are to be accurate enough for the differences which were sought to be large compared with experimental error, the absorption of the sample at the lowest frequency for which the rooms are large enough to give valid

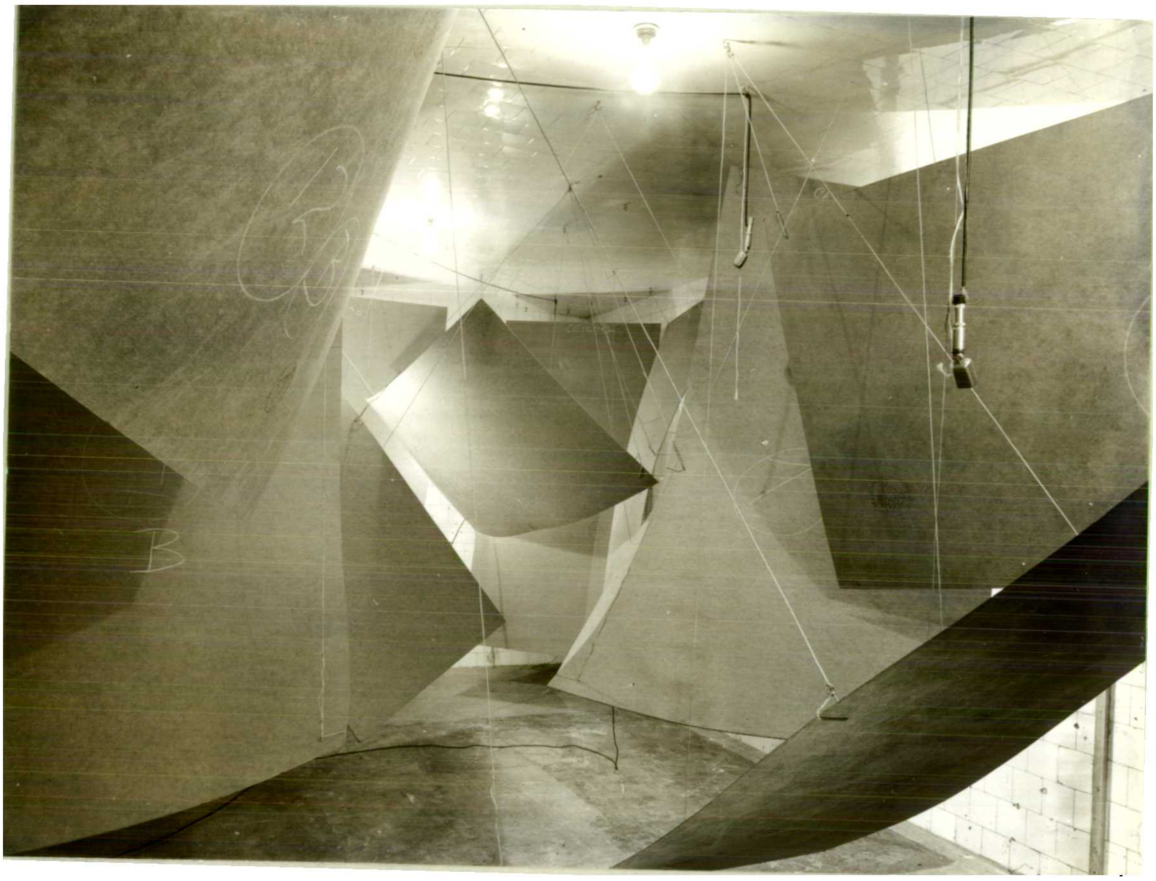


Fig. 7.3 Diffusing Sheets in Reverberation Room

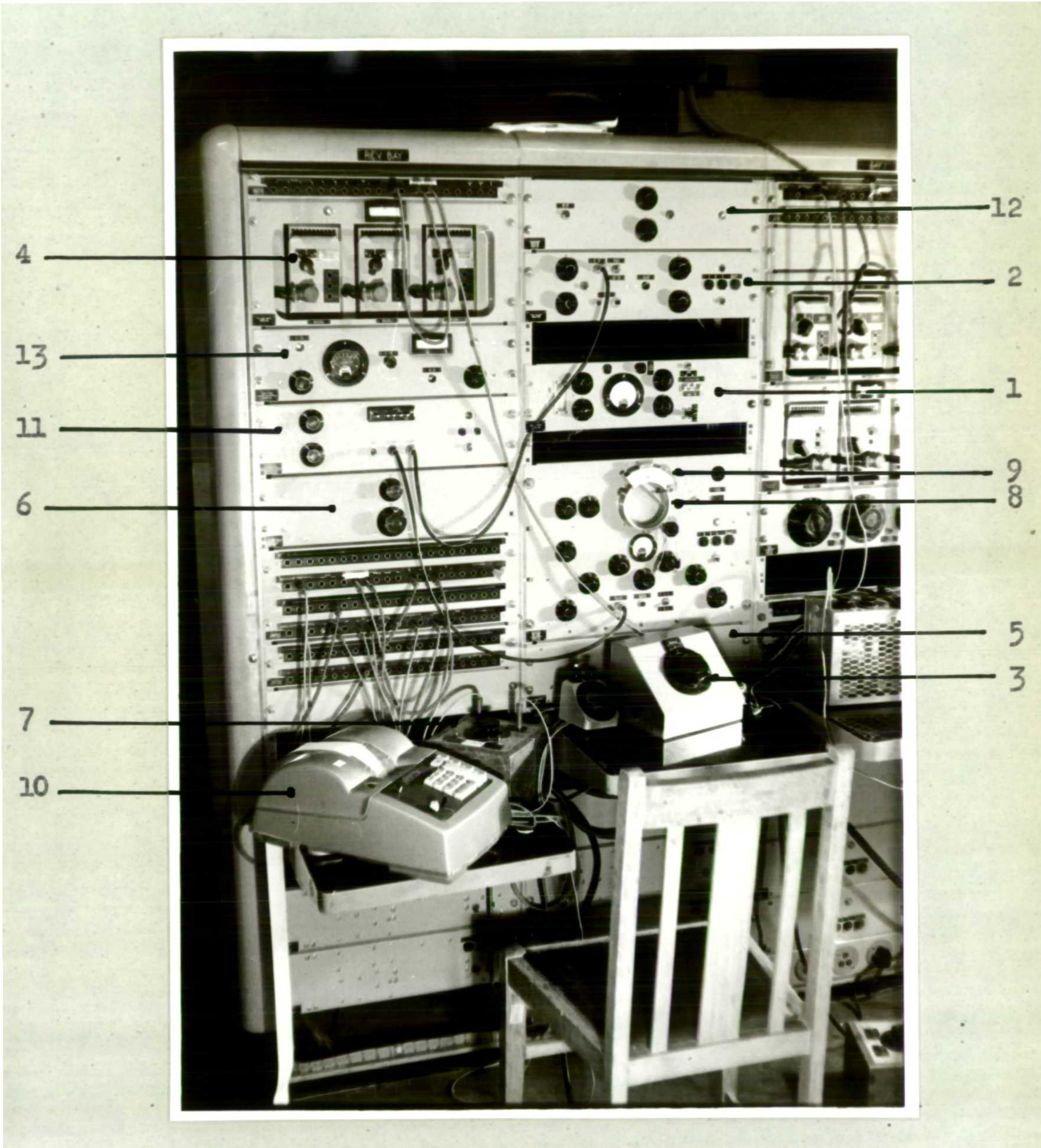


Fig. 7.4 The Measuring Apparatus

1. A.F. Oscillator
2. Tone Pulser (Freq.Mod.Tone)
3. Sequential Switch
4. Microphone Amplifiers
5. Line Amplifiers
6. Octave Band-Pass Filter
7. Attenuator
8. Oscilloscope with Logarithmic Amplifier
9. Direct reading R.T. scale
10. Adding Machine
11. Staircase Generator for Calibration
12. Valve-Characteristic Logarithmic Amplifier
13. Peak Level Meter

results* must be not too small compared with the total absorption of the surfaces of the room. With the latter consideration in mind, an absorber was chosen consisting of approximately 1" of cellulose acetate fibre and 1" of cellulose tissue, either of which materials can be placed against the wall. Fig. 7.5 shows the absorption coefficient of this material with (a) the paper tissue and (b) the cellulose acetate fibre towards the wall, measured in comparatively large areas totalling 10 m^2. The linear dimension of a sample which can be regarded as near to a corner or an edge at any frequency is inversely proportional to the frequency. Thus the maximum possible area of the sample is inversely proportional to the square of the frequency. This enabled the optimum size of sample to be roughly calculated.

Four of the samples made for the measurements are shown in Fig. 7.6. Each unit consists of a timber frame enclosing an area of 15 in. (22.8 cm) square in which the experimental material is mounted between two removable squares of flat steel wire mesh 2.5 cm apart. Two adjacent sides of the frame are 0.8 cm wide and the other two are made from 2.5 cm square section timber from which half the section is removed along a diagonal, giving a face at 45^{o} to the plane of the frame. One face of each frame is provided with two pegs projecting at right angles and the other with two holes of the

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* British Standard 3638 (1963) recommends a lower frequency limit given by $f_{min} = 125 \ (180/V)^{1/3}$ c/s where V is the volume of the room in m^3. In practice, useful results can be obtained from about one half the frequency.

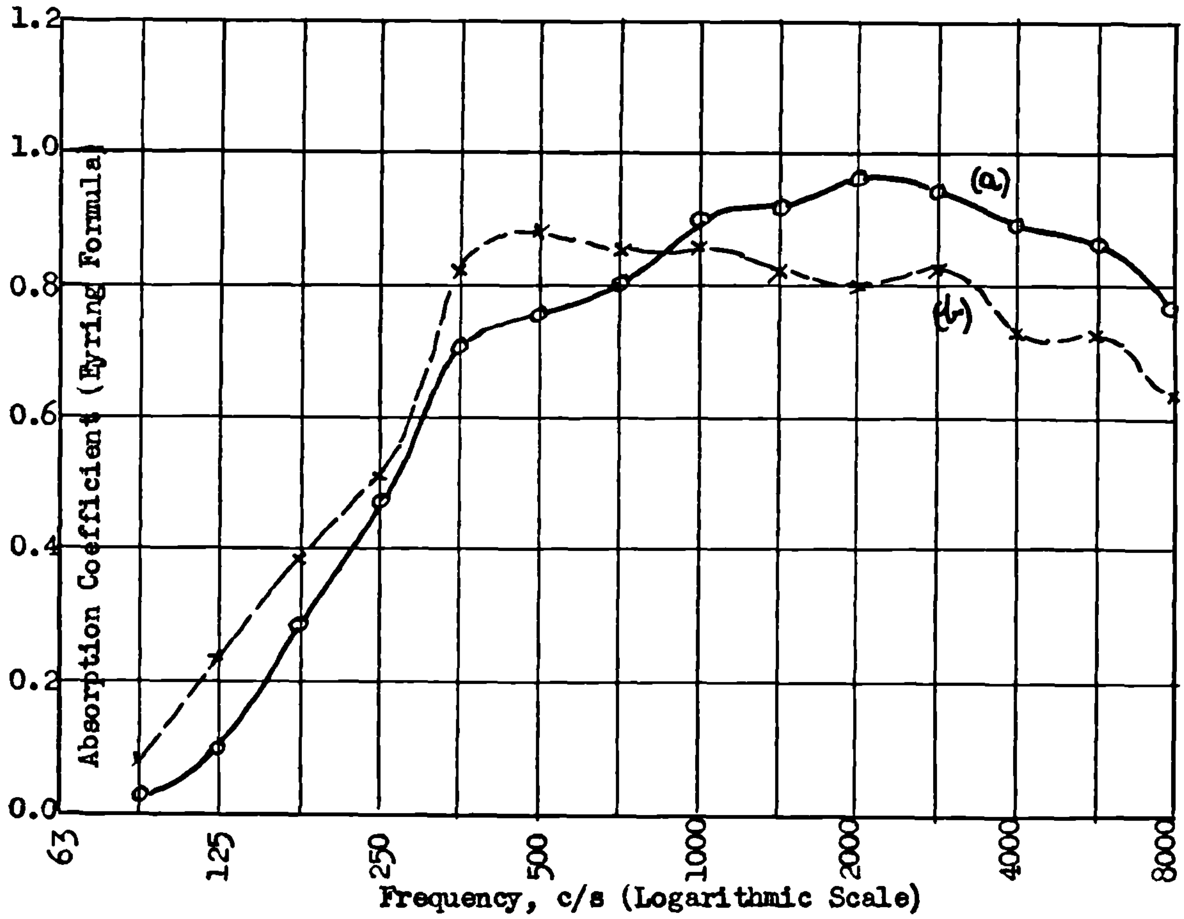


Fig. 7.5 Absorption Coefficient of Experimental Material Mounted against wall; four sheets, each 1.83 m x 1.2 m.

(a) Paper tissue against wall
(b) Cellulose fibre against wall

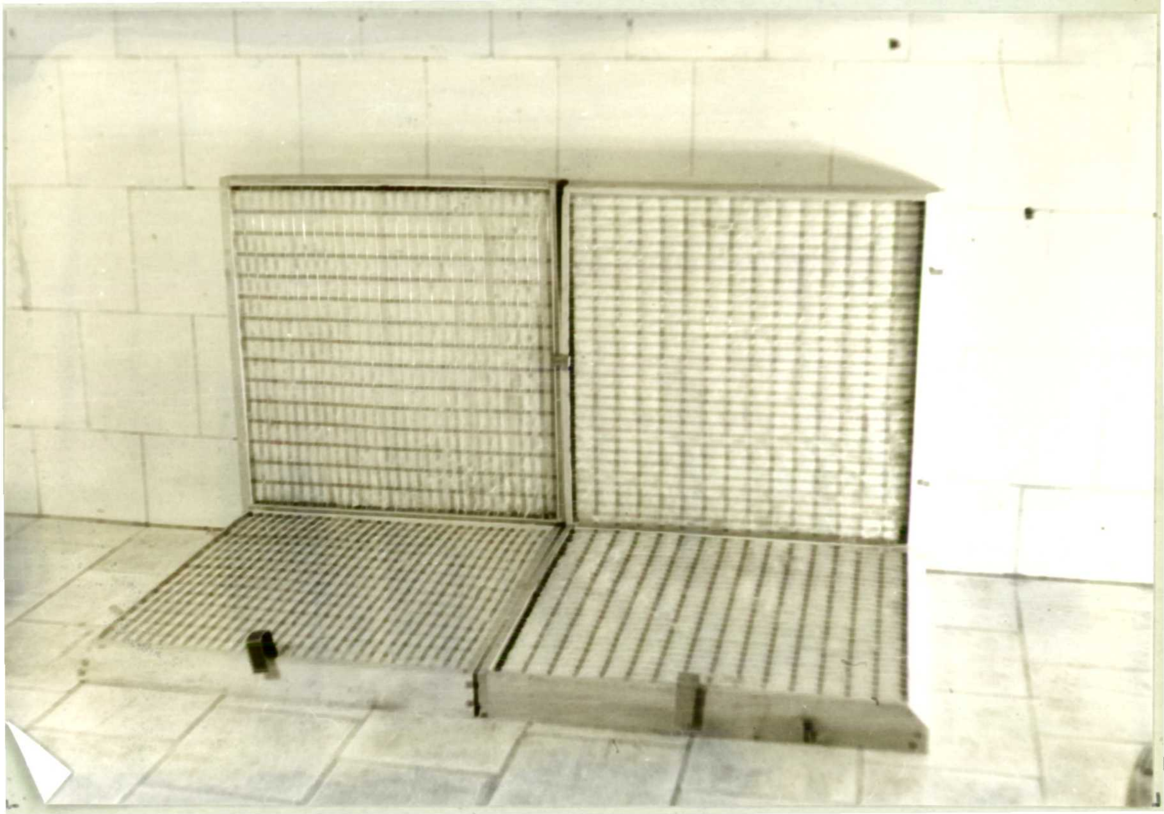


Fig. 7.6 Four Sample Units, mounted in edge of room.

same diameter. By fitting pegs into holes, three frames may be fastened together mutually at right angles and mounted in a corner of the reverberation room. Similarly two only may be fitted along an edge of the room.

In considering the arrangement of the samples in three principal positions in the room, thought must be given to the effect of the edges of the samples on the total absorption of the sample. Kuhl (1960) has shown that, except for very small samples, the effect of the diffraction at the edges is to produce an increase of the total absorption of the sample proportional to the length of the edges. The absorption coefficient will therefore be increased by an amount proportional to the ratio of the length of the edge to the area. An effort should be made to keep this ratio the same for each of the three principal positions.

There is only one possible arrangement in a corner of the room, using three square samples. If we take the length of a side as the unit the ratio of edge length to area is $\frac{6}{3}$, i.e. 2.

The same ratio is achieved by using four square samples arranged in a square on a wall face or as in Fig. 7.6 at an edge since the sample edge length is 8 units and the area 4.

These three arrangements were therefore adopted for all the measurements. Fig. 7.7 is a diagram of the reverberation room showing the sample positions.

Measurements were made of the impedance components of the material. The direct-current flow-resistance of each of the two components was measured by means of apparatus designed by the author

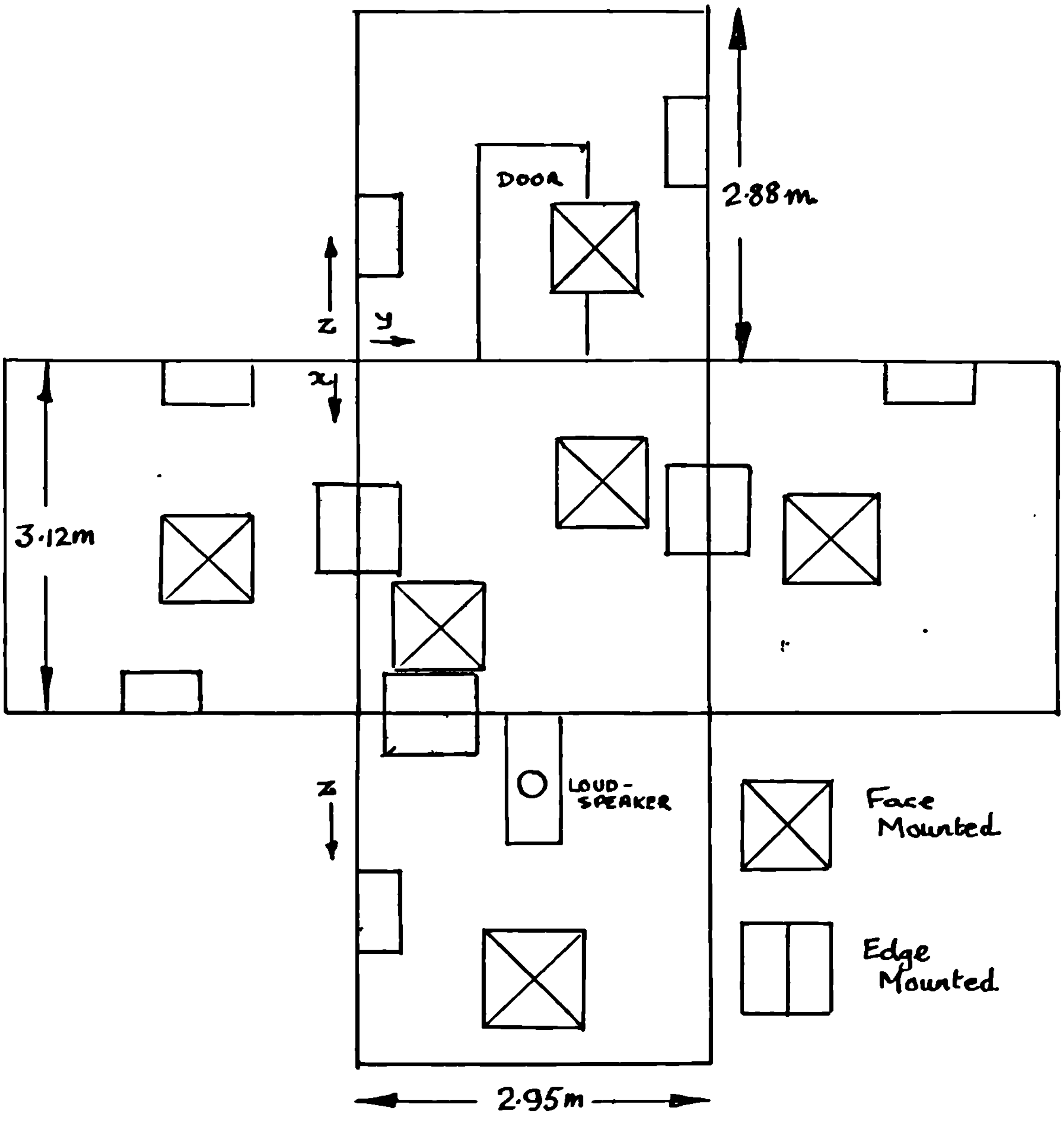


Fig. 7.7 Arrangement of Samples in Reverberation Room (Approx. to Scale)

The corner samples occupied all eight corners.
The first positions are shown, not those of the Replication Tests

Gilford (1955). Briefly, a small pump draws air at a predetermined rate of up to 100 cm^3/sec, through a sample of the material 10 cm^2 in area. A differential manometer consisting of two hemispherical shells suspended from the ends of a beam, the centre of which is attached to a torsion wire, is used to measure the pressure gradient in the direction of the air flow. The two hemispherical shells dip below the surface of paraffin in two dishes and tubes connect the air spaces bounded by the shells and the paraffin to points near the two surfaces of the sample. Any difference between the pressures beneath the two bells causes a displacement of the beam on which they are hung against the restoring force of the torsion wire since the pressures on the upper sides of the bells are equal and constant. One end of the torsion wire is then rotated to bring the beam back to its original position and the angle of rotation required is linearly related to the difference of pressure to be measured. The measurements were made at air-flow rates giving differential pressures comparable with moderate sound pressures. The characteristic flow resistance is the ratio of the pressure gradient across the sample to the velocity of air flow through it.

The real and imaginary components of the impedance of the composite material were measured using a standing wave tube of the type described by Scott (1946) and manufactured by Bruel and Kjaer. The apparatus is provided with two tubes of differing diameters, the larger capable of taking samples up to 10 cm diameter. To claculate the impedance of a sample it is necessary to know both the amplitude and phase of the wave reflected from the sample; this necessitates the measurement of the standing-wave ratio, i.e. the ratio of the sound pressures at the first maximum and the first minimum and the exact position of the first minimum, which will occur at about a quarter wavelength from the front of the samples where the reflected wave is a half-wavelength out of phase with the incident wave. There is therefore a lower limit to the frequency of measurement about equal to that at which the length of the tube is a quarter wavelength. The upper limit is determined by the appearance of transverse modes in the tube, which are superimposed on the wanted longitudinal wave and cause serious errors in its measurement. This limit is at a frequency for which the diameter of the tube is 0.6 of the wavelength, in this case 2000 c/s for the larger tube and 6300 c/s for the smaller.

The impedance measurements were used to derive a curve of the infinite area absorption coefficient against frequency. The conversion was carried out by procedures described by several authors.

Paris (1927) gives

$$\alpha = 8 \frac{\theta}{|Z|^2} \left\{ 1 + \frac{\theta^2 - X^2}{X|Z|^2} \text{ arc.tan. } \frac{X}{1 + \theta} - \frac{\theta}{|Z|^2} \log_n (1 + |Z|\theta + |Z|^2) \right\}$$

where the impedance $Z = \theta + iX$.

This depends on the assumption that the acoustic impedance at the surface of the sample is independent of the angle of incidence.

The values of Z, θ and X were substituted directly into the equation, or α was derived by the simpler procedures devised by Atal (1959) and by Dubout and Davern (1959) which are graphical representations of Paris' equation.

Results in the Reverberation Room

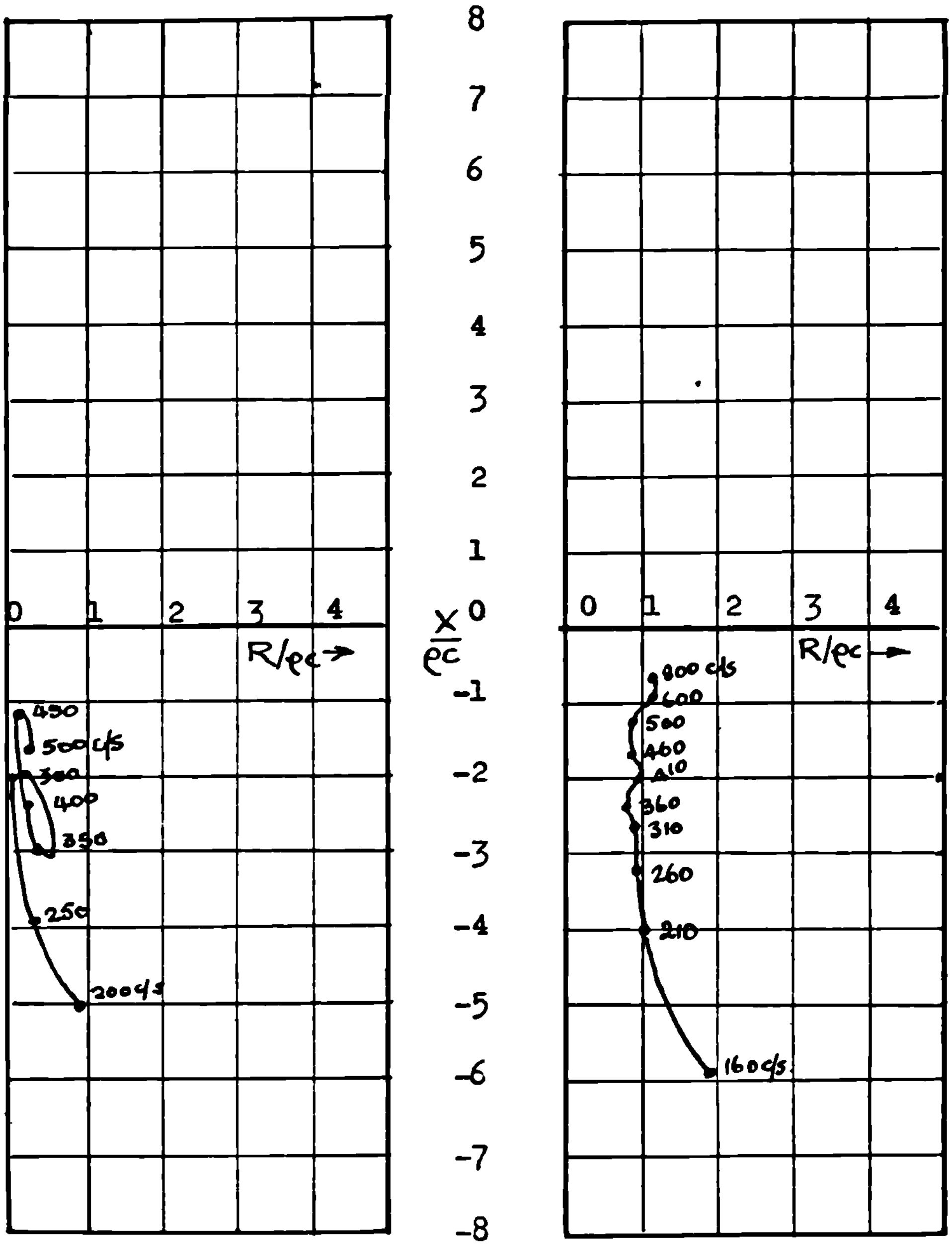
The absorber chosen for the measurements was composite, consisting of two layers of different materials of widely different impedances, so that different properties could be obtained from the same samples by simple reversal. One of the materials consisted of cellulose acetate fibres of 2.5 denier filament weight, packed together in random orientation forming a wadding similar to cotton wool. The other layer was of forty sheets of very thin cellulose tissue lightly pressed together. The cellulose acetate wadding had a thickness of 2.5 cm and a flow resistance of 7 c.g.s. units. The tissue was 1.2 cm in total thickness with a somewhat variable flow resistance, the mean of which was 90 c.g.s.u. with a standard deviation of $\pm$ 9.8 c.g.s.u. on individual samples.

Measurements were made on this composite structure, first with the tissue outwards and afterwards with the wadding outwards. To control the form of the samples, they were enclosed in the wooden frames described above and slightly compressed between the wire mesh panels above and beneath.

Initial tests were made in the impedance tube as stated above to determine the random incidence absorption coefficient in the range of frequency with which the investigation was concerned.

Figs. 7.8 and 7.9 show the impedance diagrams obtained for the materials in the two directions.

The following table gives the components of the impedances of the material in the two directions and the random incidence absorption



Acoustic Impedance Contours of Hylotex

R/ c = Normalised Resistive Component
X/ c = " Reactive Component

Fig. 7.8
Cellulose Acetate
Outwards

Fig. 7.9
Cellulose Acetate
Inwards

coefficients calculated from them.

TABLE 7.3

IMPEDANCE AND CALCULATED STATISTICAL ABSORPTION COEFFICIENT OF HYLOTEX

Tissue outwards				Cellulose acetate outwards			
c/s	R/ c	X/ c	α_{stat}	c/s	R/ c	X/ c	α_{stat}
160	1.9	-5.9	0.27	147	0.43	-0	0.0
210	1.00	-4.0	0.23	200	0.98	-5.1	0.25
260	0.90	-3.2	0.32	244	0.48	-3.9	0.30
310	0.80	-2.6	0.39	300	0.22	-2.1	0.30
360	0.78	-2.4	0.42	343	0.50	-3.0	0.23
410	0.91	-2.0	0.55	400	0.38	-2.4	0.25
460	0.85	-1.7	0.58	445	0.20	-1.2	0.32
500	0.95	-1.6	0.63	500	0.39	-1.7	0.38
600	0.90	-1.25	0.70				
800	1.10	-0.93	0.80				
1000	1.04	-0.70	0.83				

α_{stat} is the statistical or random-incidence absorption as calculated by methods given above.

To establish whether there are significant variations between the measured absorption coefficient of a material with its position in a room, it was considered necessary first to establish the extent of the variation between different determinations in one of the configurations. For the configuration in which the samples are on the wall surfaces as opposed to corners or edges, the variation could be established most generally by selecting two or more arbitrary sets of positions of the samples.

Table 7.4 shows the co-ordinates of the centres of the six groups of samples in the two sets of positions. The lower corner on the left, facing the door from inside the room, was chosen as origin.

TABLE 7.4

CO-ORDINATES OF CENTRES OF ABSORBERS FOR REPLICATION TEST (metres)

SET 1 (As in Fig. 7.7)	SET 2
(0, 2.00, 0.95)	(0, 1.4, 1.4)
(1.00, 1.90, 0)	(1.0, 1.0, 0)
(2.26, 0.61, 0)	(2.2, 1.9, 0)
(1.55, 2.95, 1.07)	(1.0, 2.95, 1.8)
(1.72, 0, 1.58)	(1.5, 0, 1.3)
(3.12, 1.53, 1.99)	(3.12, 0.6, 0.7)

The positions in Set 2 are approximate only as they were not recorded exactly. The co-ordinates 0 or 2.95 indicate the plane surface against which the sample is laid.

Fig. 7.10 shows the results of two determinations in the first set of positions and one in the second. It will be seen that the differences between the three determinations within the valid range 150 - 425 c/s fall within a total spread of about 0.05. At high frequencies the three determinations diverge by a greater amount, due possibly to differing diffusion. This will be seen more clearly in relation to later results.

The standard deviation of individual results for the three determinations is given by the following table: (p. 195).

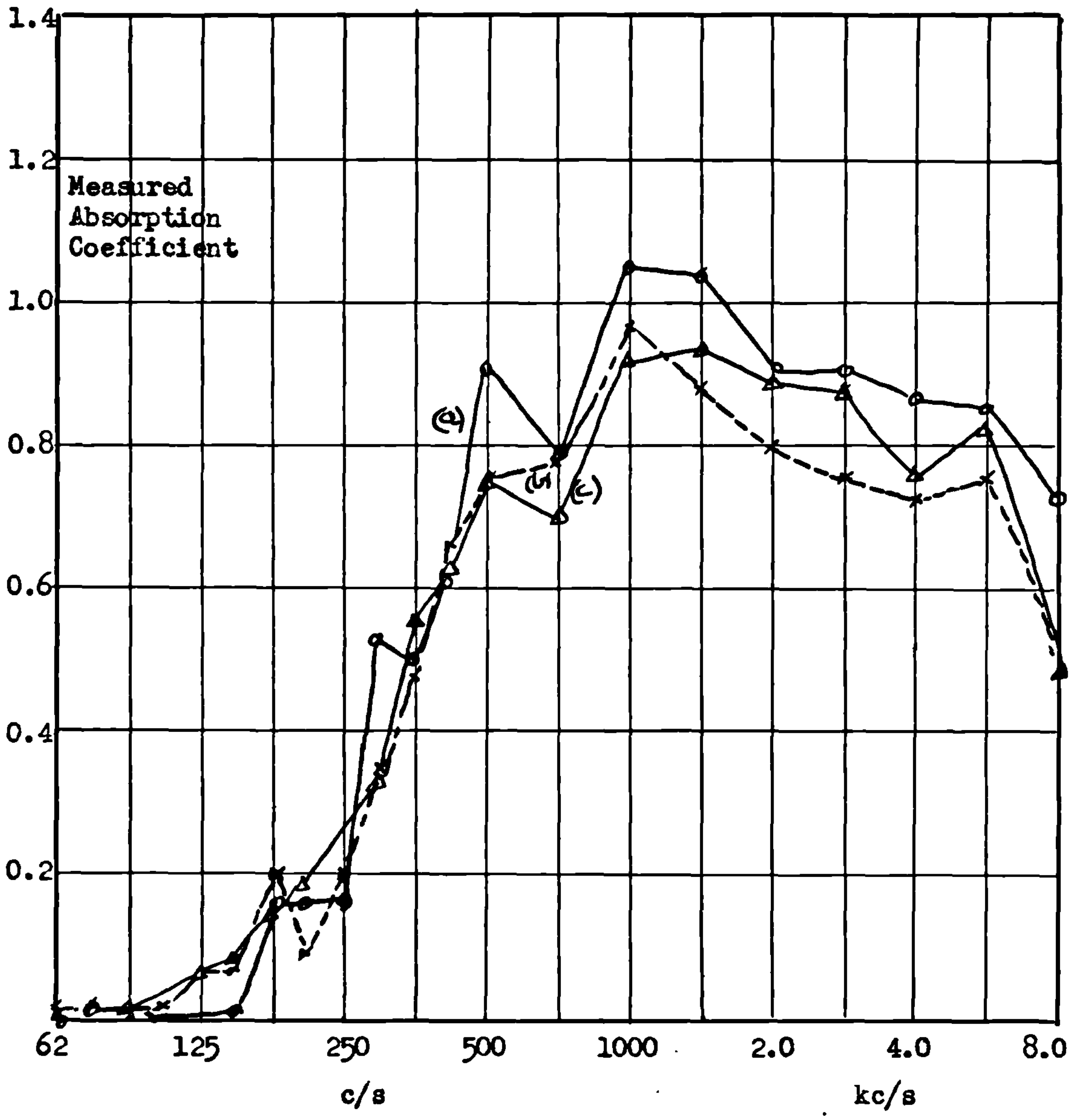


Fig. 7.10 Replication Tests. Material on Wall Faces Hylotex with Cellulose Acetate Outwards With Diffusers

(a), (b) First Arrangement. Two Tests.
(c) Second Arrangement.

TABLE 7.5

STANDARD DEVIATIONS IN ABSORPTION COEFFICIENT MEASUREMENTS

c/s	150	175	210	250	300	350	425
S.D.	0.0215	0.016	0.030	0.029	0.090	0.031	0.015

Only at 300 c/s is the S.D. greater than 0.031. At this frequency one of the original determinations is very much greater than the other which is close to the second set determination. There is a large divergence of one set from the others at 500 c/s, just outside the valid range, but this appears to be exceptional and is probably fortuitious.

The S.D. for replicates will therefore be taken as approximately 0.03 on the coefficient for a single determination.

Similar determinations for the coefficient were carried out with the samples in the corners and in the edges. The positions for the latter were as follows:

(metres)

(0, 0, 1.28)

(0, 2.95, 1.95)

(1.47, 0, 0)

(1.34, 2.95, 0)

(3.12, 0, 1.78)

(3.12, 0.80, 0)

Each arrangement was measured in the room without and again with diffusing plates of hardboard hung as described above.

Fig. 7.11 shows the results without diffusers, the cellulose

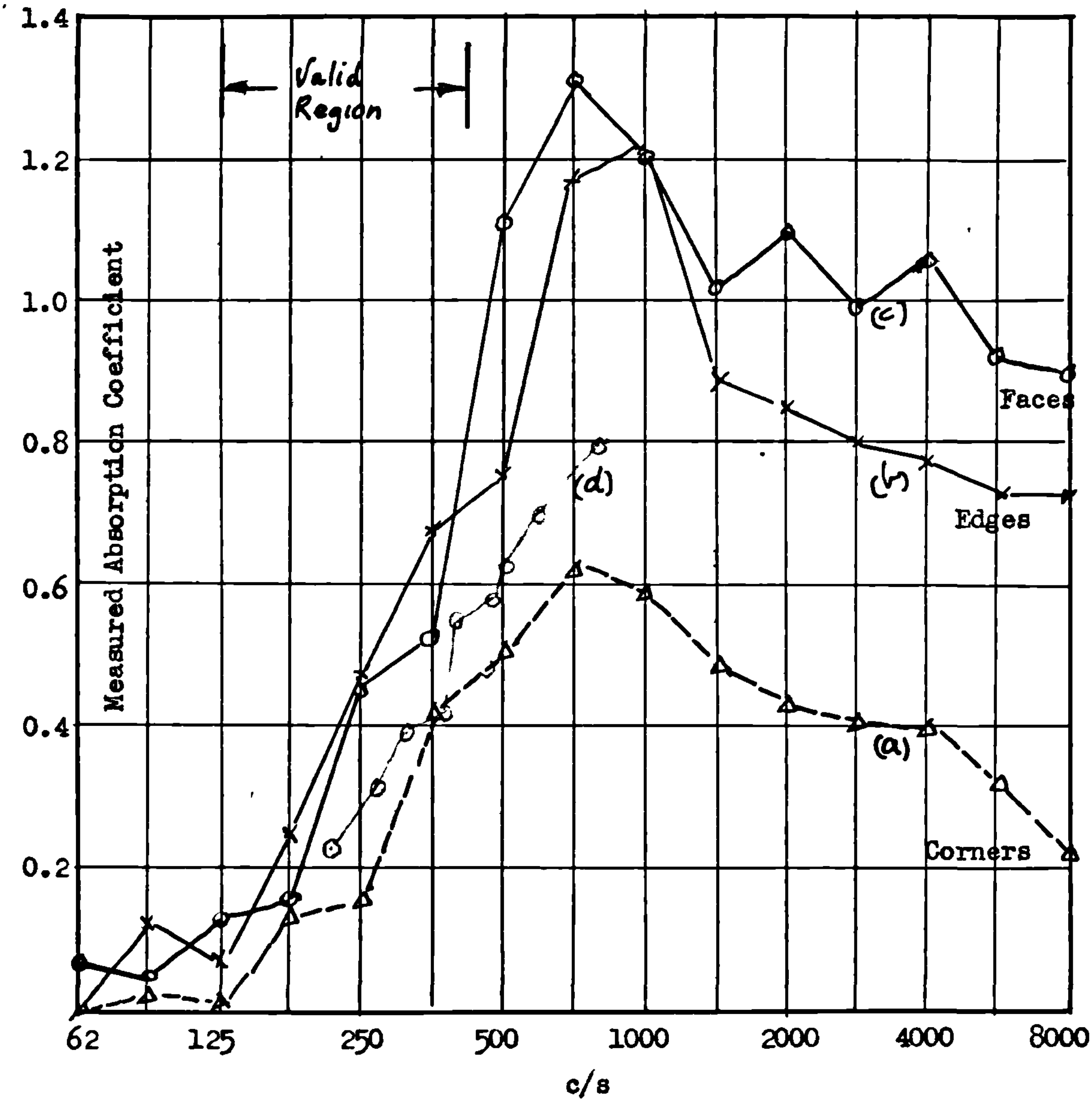


Fig. 7.11 Results for Hylotex, Tissue Outwards, No Diffusers

Material placed in (a) Corners (b) Edges and (c) on Faces of Walls

Curve (d) is α_{stat}.

tissue being outwards.

Fig. 7.12 shows the results of the same arrangement with diffusers in place.

In each figure is included a curve showing the values of α_{stat} derived from the impedance tube measurements. The valid frequency range for the experiment (125 to 425 c/s) is marked by vertical lines.

In the first of these two figures, it will be seen that the corner position gives the lowest of the three values at all frequencies within the valid range as well as outside it. The face position is next and the edges gave the highest figures.

The second of the figures, showing the results in the presence of the diffusing plates, shows a change in this order, the corners giving the highest figures within the valid range with the other two positions approximately equal.

Figs. 7.13 and 7.14 give the corresponding results for the material with the cellulose acetate layer outwards. In these figures the face positions give the lowest coefficients and with the diffusers present the corner positions give the highest. Without diffusers there is little apparent difference between the corner and edge positions within the valid range.

The differences between the positions are not very great, but may be shown more clearly by considering for each position the mean absorption in the valid frequency range, as in Table 7.6 below. (p. 201).

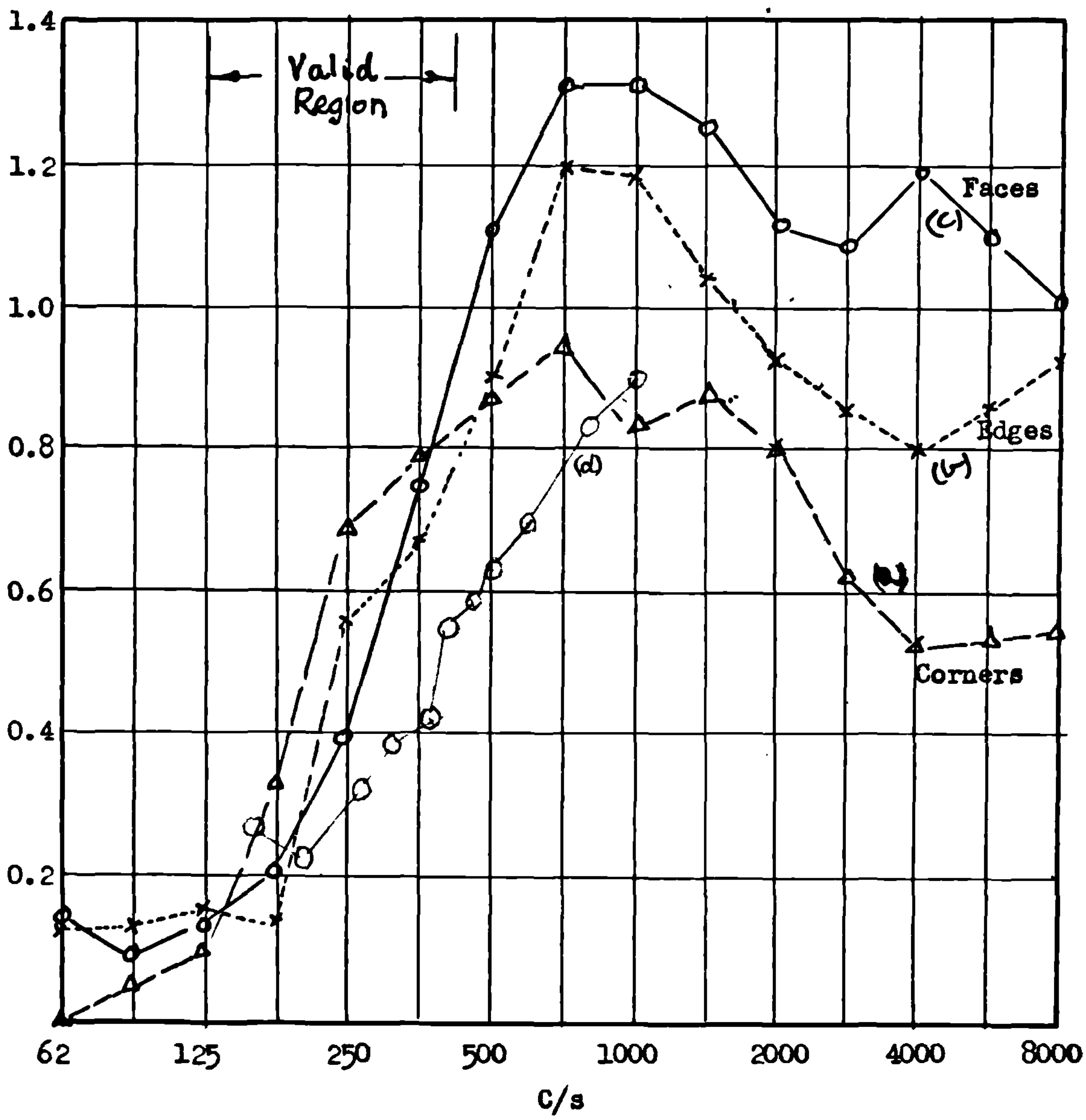


Fig. 7.12 Results for Hylotex, Tissue Outwards, with Diffusers Material Placed in (a) Corners, (b) Edges and (c) on Faces of Walls (d) is α_{stat}

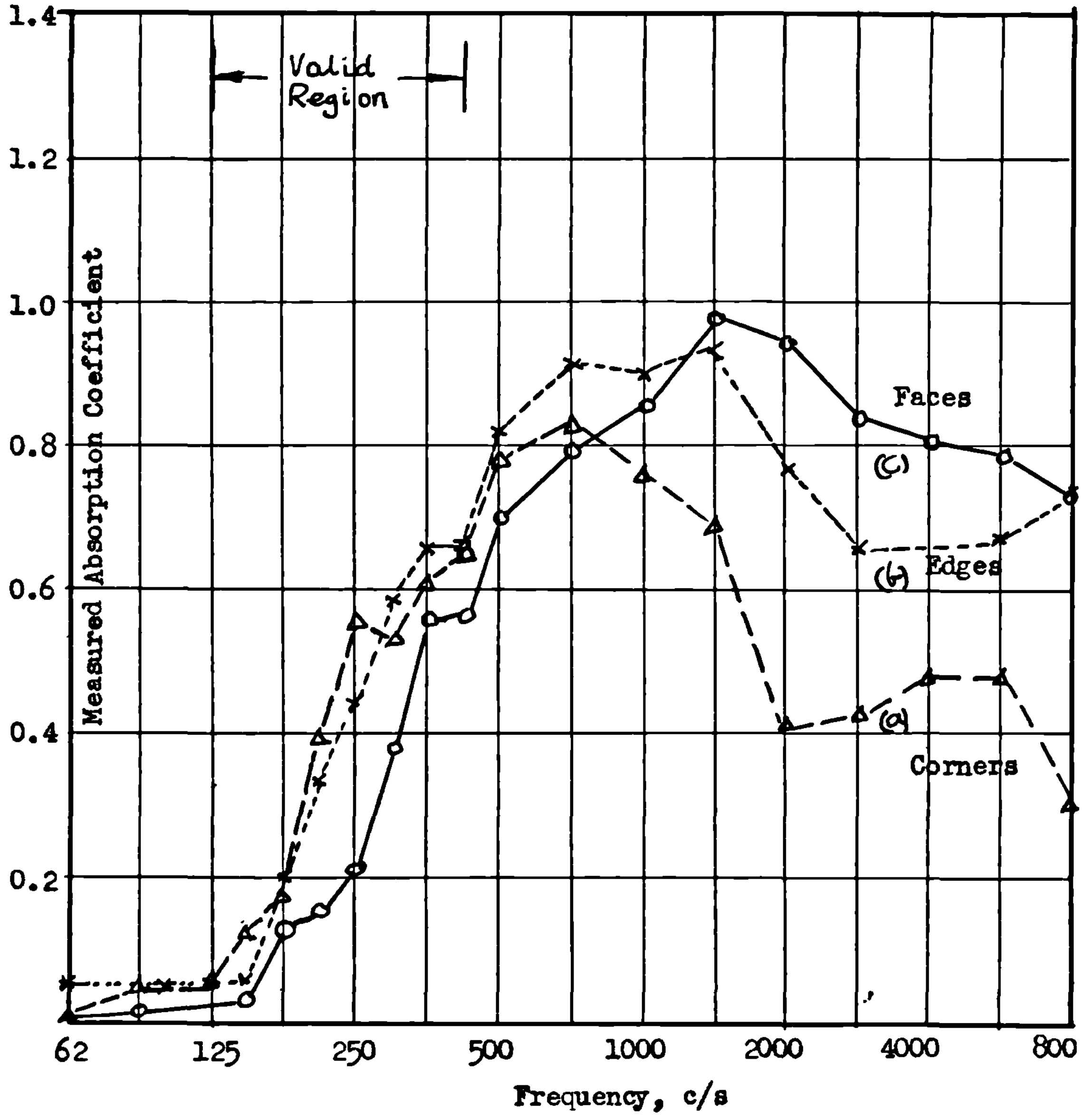


Fig. 7.13 Results for Hylotex, Tissue Inwards, No Diffusers. Material Placed in (a) Corners, (b) Edges and (c) on Faces of Walls

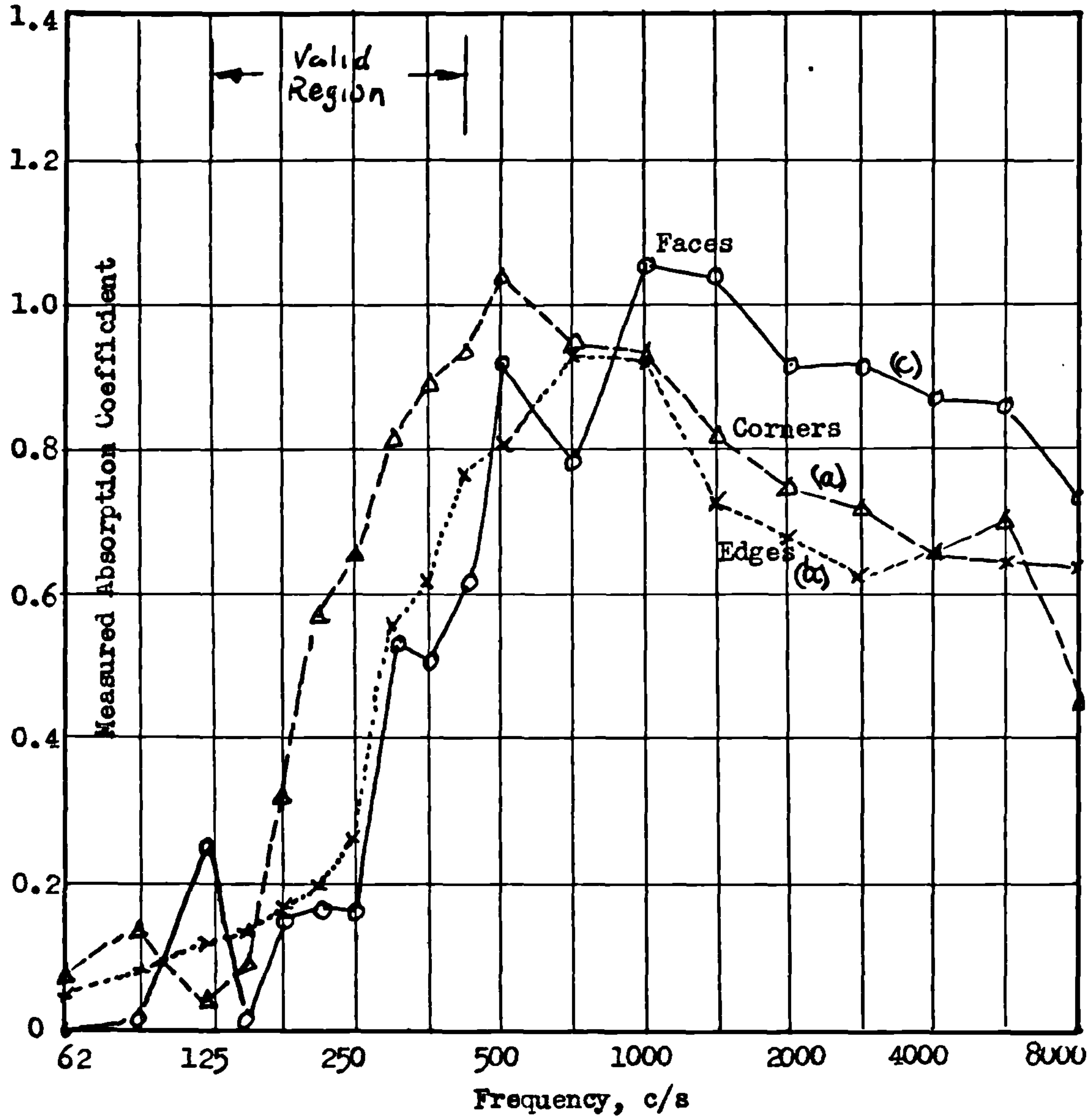


Fig. 7.14 Results for Hylotex, Tissue Inwards, With Diffusers. Material placed in (a) Corners, (b) Edges and (c) on Faces of Walls

TABLE 7.6

MEAN ABSORPTION COEFFICIENTS FROM 125 TO 425 c/s

	No Diffusers	Diffusers
1. Hylotex with tissue outwards		
Corners	0.254*	0.594
Edges	0.474	0.496
Faces	0.457	0.471
2. Hylotex with tissue inwards		
Corners	0.441	0.607
Edges	0.414	0.389
Faces	0.296	0.303

* Different frequencies used in measurements

Taking the standard deviation of a single result as 0.03, as derived above, the standard deviation for a mean of seven results is approximately $0.03/\sqrt{7} = 0.012$. The difference between the results for corners, edges and faces are therefore clearly significant except for the difference between edges and faces with the tissue outwards.

With diffusers present, the corner positions give the highest results for both directions of the material, though with the component of high flow resistance outwards the difference is must less great.

This observation agrees with the prediction that the differences should be greatest if the material has a low resistive component compared with the characteristic resistance of plane waves in air.

Without diffusers the corners show much less advantage; this is particularly so for the material with the high-resistance component in the front. The explanation must be that in the corner positions the material has negligible effect on the state of diffusion in the room, whereas when placed in the edges or on the faces, the diffusion is improved by the presence of the absorbers.

It would be expected that this phenomenon would be observable at higher frequencies than those at which the sample size becomes negligible in comparison with the wavelength, and would therefore become more important than the changes of absorption due to position.

Let us therefore compare the mean coefficients in the high-frequency region from 1 to 8 kc/s. These are summarised in the table below:

TABLE 7.7

MEAN COEFFICIENT OF HYLOTEX IN THE RANGE 1 kc/s TO 8 kc/s

		Without Diffusers	With Diffusers
1. Tissue Outwards			
	Corners	0.41	0.68
	Edges	0.82	0.94
	Faces	1.03	1.14
2. Tissue Inwards			
	Corners	0.51	0.72
	Edges	0.74	0.70
	Faces	0.85	0.91

It will be seen that in this frequency range, the corner position is the worst of the three, whether the diffusers are present or not. With the introduction of diffusers the improvement in measured coefficient is very much greater for the corner position than for the edges or faces, thus confirming the hypothesis that the diffusion produced by the samples in the corners is less than that produced by samples in the edge or face positions.

This is an important result in connection with studio design.

It will be noted that all the figures measured in the reverberation room are greater than α_{stat} calculated from the impedance tube results. The difference is very much greater with the cellulose acetate fibre outwards. It is explained by consideration of the finite size of the samples with respect to a wavelength, and has been treated both theoretically and experimentally by various authors.

The usual treatment is to regard the total absorption of the sample as being the sum of that due to the sample regarded as a portion of a sample of infinite area and of an additional absorption caused by diffraction of sound near the edges of the sample. It has been noted in Chapter 6 that Kuhl (1960) obtained an increase of maximum coefficient from 0.9 to 1.53 by subdividing a sample into rectangles or squares of 1 m^2 and smaller. The area of the wall-mounted samples in the present investigation was 0.58 m^2.

Northwood, Grisaru and Medcof (1959) have treated the edge effect theoretically, obtaining equations of considerable complexity which are difficult to apply.

We can also view the matter according to the concept of reduced radiation resistance as described in Chapter 6. If a single sheet of an absorber, matched to plane-wave radiation resistance, is dispersed uniformly in the form of small patches occupying one nth of the wall area, it may be shown that, very approximately, the effective absorption coefficient is increased in the ratio $4n^2/(1 + n^2)$. This has a limiting value of 4 where $1/n$ tends to zero.

CHAPTER 8

THE INFLUENCE OF DIFFUSION IN SMALL ROOMS

8.1 General Remarks on Diffusion

It is proposed here to give a brief statement on the influence of diffusion in small rooms and the means of obtaining it. The subject of diffusion has been touched upon in Chapters 3 and 7, and its relation to the reduction of colourations will be considered. No new experimental material is presented though the results of accumulated experience and observation during the past four years will be included.

The sound field in a room is defined as being in a state of perfect diffusion if it has uniform energy distribution throughout and if the directions of propagation at any arbitrarily selected points are wholly random. If the sound field in a room takes the form of a single standing wave pattern associated with an isolated mode, the first condition will be satisfied since the sum of the kinetic and potential energies of any elementary volume of the field will be constant. The second condition, however, will clearly not be satisfied, since the particle velocities at all points will be parallel to the same direction if it is an axial mode or to one of two or three directions if it is a tangential or oblique mode.

Any improvement in the diffusion of a small room, therefore, implies the breaking-up of strong standing wave systems, the diversion of energy from strong modes, particularly axial ones, into weaker, more oblique modes, and the retardation of the establishment of strong modes, with the consequent reduction of audible colourations.

For a very complete account of the production, measurement and effects of diffusion in small rooms, reference should be made to a paper by Randall and Ward (1960) who investigated the effect of diffusion in increasing the absorption of sound by a patch of absorbing material in the room, and the measurement of diffusion by various possible methods. The effect of absorption has also been carefully investigated by Kosten (1960) and Burd (1963) in connection with the measurement of absorption coefficients by the reverberation room method. It is found that the measured absorption coefficient of a single sample of material may increase several fold if the diffusion of the sound field in the reverberation room is made diffuse. The diffusion is normally effected by adding irregularities to the room surfaces or by hanging sheets of reflecting material in a random manner from the ceiling so as to change the direction of propagation at every reflection. Randall and Ward showed that small patches of absorber are extremely effective in improving diffusion, so that the introduction of absorbing material on one wall of a bare reverberation room increases the measured absorption of material on other surfaces. Burd's paper gives recommendations on the amount of diffusing material necessary for this purpose.

With regard to the measurement of diffusion, Randall and Ward finally reject methods depending on the irregularity of decay curves, which they find of more value in large enclosures such as Concert Halls (Somerville 1953, Somerville and Gilford 1957). They find the most reliable and easily derived measure of diffusion is the change of mean slope of the logarithmic display of the decay of reverberant sound pressure in the room after the cessation of a pure tone. This change of slope is well understood as the consequence of the presence of several modes with differing decay rates. It therefore has a very close connexion with the audibility of colourations.

The audibility of an axial mode as a colouration is mainly dependent on its isolation from neighbouring modes. In a very diffuse room, effective isolation may be prevented by the presence of other axial modes equally strongly excited within a frequency interval smaller than or comparable with the bandwidth; at higher frequencies, as already mentioned, it is prevented by the masking of axial modes by the contribution of great numbers of tangential and oblique modes which are individually of no importance.

If the damping of one set of axial modes is substantially less than that of neighbouring modes, as, for example, when one pair of opposite walls in a rectangular room is devoid of absorbing treatment, three effects occur. Firstly, the bandwidths of these modes is less than that of more highly damped ones, thereby making isolation more probable; secondly they will reach high standing-wave ratios; and thirdly they will persist after the neighbouring modes have decayed into inaudibility. A room with this fault shows a harmonic series of colourations to a very much higher frequency than the normal limit of approximately 300 c/s. In some parts of the room it will be possible to hear flutter echoes in which the timbre of a short impulsive sound is accurately reproduced in a train of reflections between the two opposite faces. If two pairs of surfaces are untreated, the subjective result will be a number of distinct colourations or "rings" some of them at quite high frequencies.

8.2 Reduction of Colourations by Diffusion

The first step in designing a small room such as a talks studio is to ensure that each of the three pairs of walls has substantially the same mean absorption coefficient at all frequencies, particularly those within the colouration region. The author (1959) has given the general rule that the ratio between the mean absorption coefficients of any two pairs of parallel surfaces should not exceed about 1.4:1. This is sufficient to ensure that all the modes have approximately equal decay times. If one dimension of the room is very much greater than the others, the tendency should be for the surfaces at right angles to this dimension to be more generously treated, as otherwise the decay times of axial mode in this direction will be unduly long owing to the greater mean free path.

It is often stated that to make a room non-rectangular will improve the diffusion, but the theoretical and experimental evidence is conflicting. Nimura and Shibiyama (1957), working with models, showed that the irregularity of the transmission characteristic, (steady-state frequency characteristic), was reduced by altering the angles of the walls up to 5° divergence, so as to avoid parallel pairs. They took this to be evidence of improved diffusion. However, Schröder (1954) has shown that the irregularity depends only on the volume and reverberation time of the enclosure. The results may be explained by the fact that splayed walls will discourage the establishment of repeated grazing reflections and thus tend to improve the efficiency of the absorbers, reducing the reverberation time.

Schubert and Steffen (1961), however, showed that a rectangular enclosure gives a more regular distribution of modal frequencies than a non-rectangular one and would thus be expected to have better diffusion and less prominent colourations. In the author's experience there appears to be no decisive advantage either for rectangular or non-rectangular shapes. This paper was more fully reviewed in Chapter 2.

The commonest methods of improving diffusion, as already mentioned, is to create irregularities in the wall surfaces which give scattered reflections. The dimensions, including the depth, of such irregularities must be comparable with the wavelength if they are to be effective and it has been shown by Somerville and Ward (1951) that effects can be observed if the depths of the irregularities are more than about one seventh of the wavelength. We are here concerned with the elimination of colourations occurring at frequencies down to, say, 80 c/s. The wavelength at this frequency is 14.1 ft (4,3m), requiring irregularities of the order of 2 ft (61 cm) in depth. In a small room this would be extremely wasteful of space and also unsightly; for this reason the author favours the use of patches of absorber to create the necessary diffusion.

The absorbers are constructed in units of various sizes and distributed over the surfaces with spaces in between. It is not necessary for them to be of a depth of one seventh of a wavelength or more since scattering is obtained from the diffraction of sound into the absorbers, particularly with the low-frequency membrane types. Care is taken to distribute the absorbers among the three pairs of parallel surfaces so as to yield similar mean coefficients at low, middle and high frequencies.

Even moderately-sized untreated areas facing each other directly across the room are avoided as these may cause flutter echoes, and regularities of pattern on the various surfaces are likewise avoided as they appear to cause colourations at high frequencies which are difficult to cure.

As an example of the importance of great care in this respect, Fig. 8.1 (a) shows the reverberation curve of a small effects studio in Belfast, designed for an upper-frequency reverberation time of 0.27 sec., which was investigated by the author, while curve (b) shows the curve after correction of poor diffusion caused by two opposite untreated surfaces, one amounting only to 1.3 m^2. No extra absorbing material was introduced, the cure being effected by a change in the positions of existing absorbers. The reduction of reverberation mainly at high frequencies is entirely a result of the improvement in diffusion.

The application of these methods, together with the scientific use of absorbers as described in the last two chapters has, within the last five or six years created a general improvement with respect to colouration in the B.B.C. studios.

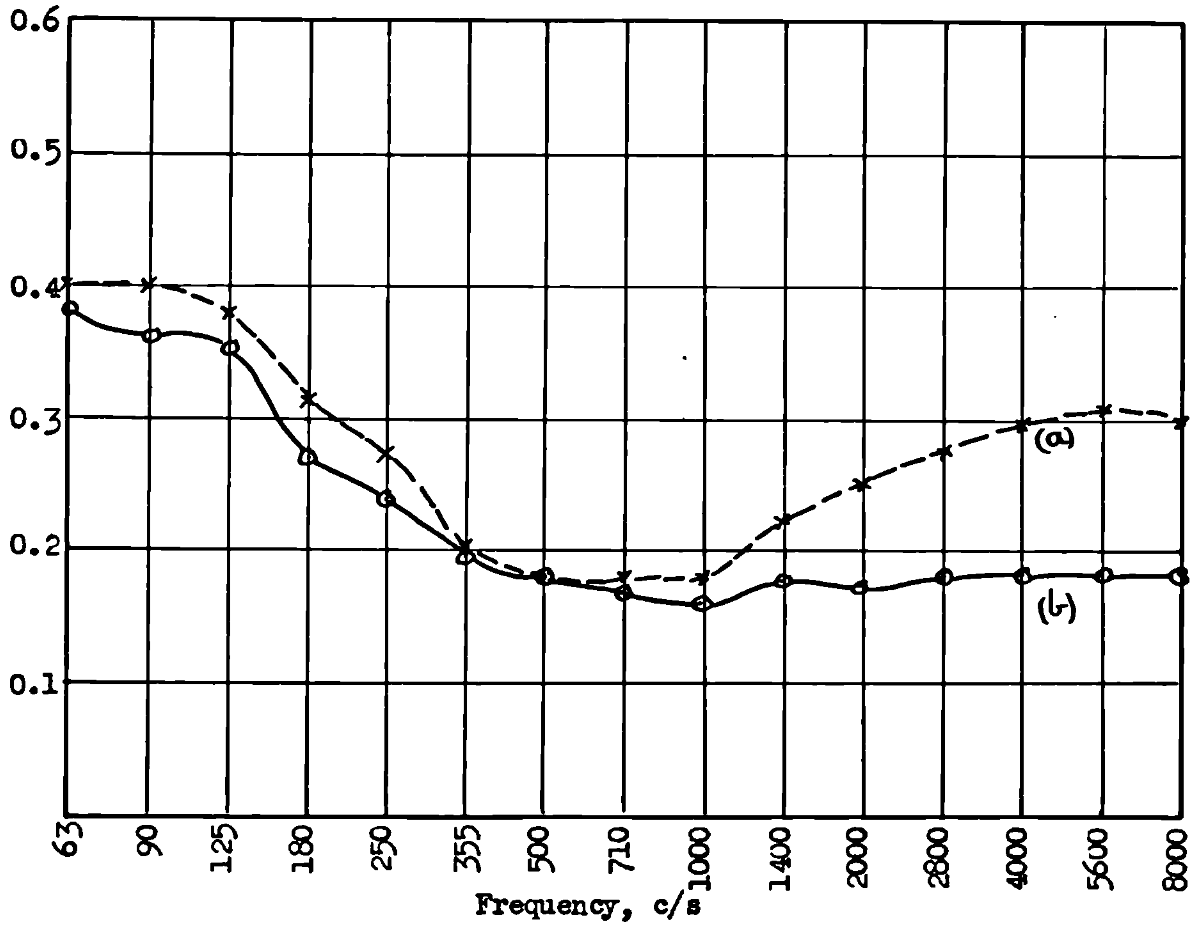


Fig. 8.1 Effect of Diffusion on Reverberation Characteristic of Small Studio

(a) Original Condition
(b) After Improvement of Diffusion

CHAPTER 9

CONCLUSIONS

The main conclusions from this work are as follows:

(1) The methods of prediction of colouration frequencies in small rectangular rooms, as postulated by the author in a previous paper (1959) which is bound with this thesis, have been borne out by further experience, though it has not been possible to improve the proportion of successful predictions.

(2) An attempt has been made to extend image-space concepts to nearly rectangular rooms, and thus to establish similar criteria for the relative importances of different classes of mode. As this was intractable, experimental work was carried out using models of thin rectangular and non-rectangular rooms. Surprising similarities were found between models of these different shapes, the mode sequence up to high orders agreeing very closely.

A small systematic difference between calculated and measured mode frequencies for the rectangular model remained unexplained in spite of considerable study and experiment.

(3) The various methods of displaying and assessing colourations are reviewed and construction of a speech spectrograph in which the spectrum from 80 c/s to 300 c/s is divided into bands of 10 c/s width and displayed appears to be the most promising method. The apparatus is under construction. The problems of filter design which are here reviewed have been solved and some have been constructed. The remainder of the equipment is not yet complete.

(4) The theory of the reduction of colourations by absorption has

been re-examined. It is concluded that, without infallible methods of frequency prediction, the best method of prevention in new designs is to provide adequate continuous low frequency absorption over the whole frequency range affected. This is most conveniently carried out by membrane type absorbers which are ideally suited for this purpose, as they possess a suitable bandwidth and can be adjusted to absorb in any desired part of the range.

Helmholtz absorbers could be used for the same purpose but are less simple to apply.

A theoretical treatment has been carried out for so-called 'functional' absorbers which are solid objects intended to be hung in free room-space instead of being mounted on the surfaces. Considerable claims have been made for their efficiency but the analysis shows that they have no special advantages. A method of using functional absorbers for narrow-band absorption is indicated but it was not further pursued since the theoretical basis is obscure and it does not appear to be of any value in the suppression of colourations.

(5) The other method of suppressing prominent modes is by the use of Helmholtz resonators of very narrow bandwidth. This can be applied only if the frequency of the mode is exactly known and measurements can be made in the untreated room. For this reason this method can be regarded only as a remedy, not as a preventative.

Other types of absorber are unsuitable for remedial selective absorption as they are essentially too great in bandwidth, which is related to their physical dimensions. The possibility is established of making membrane units of small areas which would have a small enough bandwidth but there would be practical disadvantages.

(6) The effect of the position of the various types of sound absorber in a room has been examined since it is often suggested that corner or edge positions where the sound pressure is greater than in the central regions of the wall surfaces would have advantages, increasing the efficiency of the absorbers.

This question has already been answered by Wöhle for Helmholtz absorbers, in which the only area presented to the sound-field is the cross-section of a small hole. Here the maximum absorption which can be obtained from a resonator of given volume is less altered by its position in the room than might be expected from the changes in pressure. However, if the resonator is matched for maximum absorption its bandwidth will be doubled for an edge position and quadrupled for a corner. It would therefore be more efficient as a general low frequency absorber. Since only a small area of the room surface can be regarded as being in a corner or edge, however, the advantage to be obtained by this device is very small except at very low frequencies.

Approximate theoretical treatment for membrane absorbers shows that there may be a small increase in absorption at frequencies up to about 150 c/s if absorbers of typical size are mounted in corners.

The effect of position on porous absorbers was investigated by experiment since theoretical analysis is too difficult. With absorbing materials of two different flow resistances, increased absorption was obtained at low frequencies in edges and corners compared with positions in the central regions of the walls, but at high frequencies the corner positions gave very low measured coefficients and edge positions gave coefficients intermediate between the corner and wall face positions.

This result is attributed to the absence of diffusion by the corner or edge-mounted samples.

(7) The effects of diffusion have been re-examined and recommendations are made for the application of diffusion to the reduction or prevention of colourations. The most important measures are to equalise the mean absorption coefficients of the three pairs of wall surfaces and to avoid untreated surfaces opposite to one another, even if they are quite small in dimensions.

CHAPTER 10

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THE ACOUSTIC DESIGN OF TALKS STUDIOS AND LISTENING ROOMS

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THE ACOUSTIC DESIGN OF TALKS STUDIOS AND LISTENING ROOMS

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SUMMARY

e paper gives the current views of the author and his colleagues e Engineering Department of the British Broadcasting Corpora- on the design and construction of talks studios and listening rooms ntrol cubicles, which are considered together on account of their arity with respect to acoustic behaviour. It is shown that a ictive characteristic is that, because their dimensions are com- ole with the wavelength of low-frequency sound, the sound field aracterized by strong simple standing-wave patterns which cannot iminated without eliminating the reverberation itself. It is shown that the audible effects are confined to those associated with le axial modes and that, by careful adjustment of dimensions, sion of diffusion and the proper distribution of absorbing material, orst faults can be avoided. The effects of the monaural listening are considered as well as the consequent necessity for reduced ground noise and reverberation in studios as compared with a al living-room.

ally, design data for both talks studios and control or listening les are given.

(1) INTRODUCTION

ere are over 120 B.B.C. studios used for talks, news, ssions, or continuity announcements. There are also about acoustically treated rooms used for the control and moni- g of programmes, quality checking and similar purposes. hese have one common feature: they are comparatively l, with volumes between 30 and 120 m^3.

their acoustic behaviour there are two main differences een large and small studios. The most fundamental is that, eas the wavelength of sounds in the low-frequency end of udible spectrum is of the order of, or even greater than, the nsions of a small room, a very large studio has dimensions compared with all but the very longest relevant wave- hs. Consequently it may be shown that such a small will have clearly defined resonances characterized by g standing-wave patterns which are not altered in their al nature by small changes in the room shape, whereas resonances are usually absent from larger buildings. In of clearly defined frequency effects, however, large buildings have echoes which are characterized by their time delays and which can be influenced by small changes in the shapes of the walls and ceiling.

The second difference, mainly practical but only slightly less fundamental, is that the reverberation times associated with large studios are generally longer than those of small studios. This is partly because with surfaces of a given average absorption coefficient the reverberation time is proportional to the cube root of the volume, and partly because the optimum reverberation time for a large studio is generally greater than that for a small one. In a small studio the time may be so short that reverberation is no longer appreciable as a time-extension of the original sound, as it is in an orchestral studio or a concert hall, but only as an alteration of the frequency content, making speech, for instance, sound more 'bassy' or more 'sibilant'.

To summarize, large studios are characterized by reverberation and echoes with recognizable time-scales, and small ones mainly by phenomena recognizable as frequency effects. These differences result purely from the disparity in size, and small-studio problems are shared by all small rooms for which good acoustics are required. It seems appropriate, therefore, to consider small studios and listening rooms together in a single paper, first dealing with their common acoustic properties and later describing the detailed treatment required to make them suitable for their different uses. The term 'listening' room' will be used throughout to mean any room such as a quality-checking room, control cubicle or control room which requires good listening conditions but is not used as a studio.

(2) GENERAL ACOUSTIC PROPERTIES OF SMALL ROOMS

(2.1) Formation of Simple Modes in Small Rooms

It has been stated above that the most important characteristic of a small room is the fact that its dimensions lie near or within the wavelength range of low audible frequencies and that this gives rise to recognizable resonance effects known as colorations. If we consider first the simplest possible type of 'mode' or standing-wave system, which is formed when plane waves are reflected between a pair of opposite walls parallel to each other,

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the lowest frequency at which a standing-wave pattern will be formed is given by the quotient of the velocity of sound and twice the distance between the walls. At this frequency there will be pressure antinodes at the boundary surfaces and a nodal plane half-way between them. Multiples of this fundamental frequency will give modes with antinodal planes dividing the distance between the boundaries equally into two, three or more equal parts. The frequencies of these modes are given by

$$f_n = \frac{nc}{2l} \quad . \quad . \quad . \quad . \quad . \quad . \quad . \quad (1)$$

where f_n is the nth harmonic of the fundamental, l the relevant room dimension and c the velocity of sound.

There will be room modes of this kind in all parts of the audible spectrum, but whether or not they will be appreciable as colorations depends on the following factors:

(*a*) The bandwidth of the mode.
(*b*) The degree of excitation of the mode.
(*c*) Its separation from neighbouring strongly excited modes.
(*d*) The positions of the sound source and microphone, with respect to standing-wave systems.
(*e*) The frequency content of the source.

Since the colorations largely determine the sound quality from a studio or the goodness of a listening room, the next two Sections will be devoted to an examination of these five factors.

(2.2) Bandwidth of a Room Mode

The ratio of the pressure at a modal frequency f_n to that at any other frequency f may be shown to be

$$\frac{p_{max}}{p_f} = \frac{1 - 2r^2 \cos(4\pi fl/c) + r^4}{(1 - r^2)\{1 + [1 + \cos(4\pi fl/c)]\}r + r^2} \quad . \quad (2)$$

where r is the mean reflection coefficient of the walls.

The derivation of this equation, which follows without difficulty from the image-plane concept described in Section 2.3, will be omitted for reasons of space.

Defining the bandwidth conventionally as the frequency difference between points on either side of f_n at which the pressure has fallen to $1/\sqrt{2}$ of its peak value, we find:

$$\text{Bandwidth} = \frac{c}{2\pi l} \text{arc cos} \frac{(1 + r)^4 - (\sqrt{2})(1 - r)(1 - r^3)}{(\sqrt{2})(1 - r)^2 + 2r^2}$$

$$= f_1 \phi(r) \quad . \quad . \quad . \quad . \quad . \quad . \quad . \quad . \quad . \quad . \quad . \quad (3)$$

where $\phi(r)$ is a function of r alone.

It is clear that the bandwidth depends only on the reflection coefficient of the walls and the frequency of the fundamental mode. In a room having a reverberation time independent of frequency, therefore, all harmonics of a given mode will have substantially the same numerical bandwidth. Such a room will thus have series of modes of comparable bandwidth over the whole audible range, the narrowest bandwidths being associated with the longest room dimension. A typical value for a small studio is about 5 c/s.

(2.3) The Relative Importance of Axial, Tangential and Oblique Modes

So far we have considered only simple 'axial' modes formed by reflection between two parallel wall surfaces. Two other classes are possible: those formed by reflection between two pairs of surfaces, known as 'tangential' modes, and those involving all three pairs of surfaces, known as 'oblique' modes.

The relative subjective importance of the three classes has been dealt with in an important paper by Mayo,[1] who calcula rates of build-up of reverberant sound pressure from indivic images and groups of images arranged in lines and pla Fig. 1 is a two-dimensional representation of the images forr by a point source in one corner of a room, the spacing in

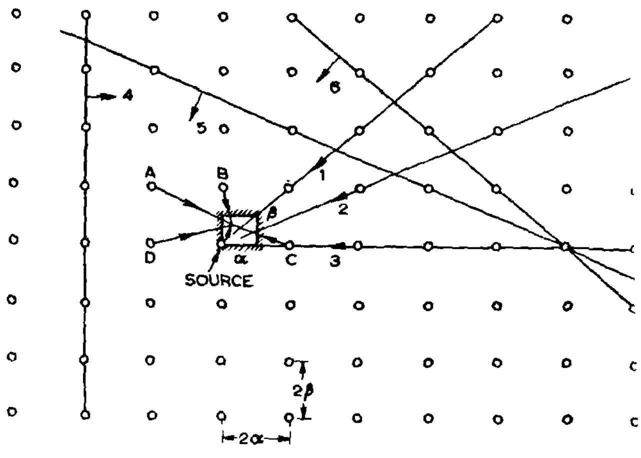


Fig. 1.—Arrangement of images in one plane surrounding a rectangu studio, showing random reflections from images A, B, C, D, l arrays 1, 2, 3, and plane arrays 4, 5, and 6.

two directions being twice the corresponding dimensions of room.

Mayo considers the way in which the sound pressure at point in the room builds up when the sound source in the corr is suddenly started. Simultaneously with the start of the sour all the images appear and start to radiate as point sources, t sound from them reaching the room in order of their distanc Each image gives rise to radiating spherical wavefronts, t pressure diminishing inversely as the distance. Furthermo there is a loss of pressure by absorption at each reflection, a after the nth reflection the strength of each image has be reduced in the ratio of r^n : 1, using the notation of the previo Section. Consideration of the manner in which the images a formed shows that the number of reflections is roughly propc tional to the distance from the source, so that the pressu amplitude in the room due to a particular image from whi the sound is arriving at time t may be seen to be roughly pr portional to r^{kt}/t, where the constant k is a function of the roo dimensions and the velocity of sound.

Considering first the sound from the nearest few images, see that there is no special relationship between their distanc and the wavefronts arrive at random time-spacing and in rando phase. The amplitude rises quickly since the images are clos but there is no systematic reinforcement of particular frequencie Owing to the fact that the pressure decays rapidly according the same law as that given in the last paragraph, the effect very temporary and soon gives way to a second regime, involvi groups of images arranged in rows.

If we consider the effect of a row of images lying on a li which passes through the studio, we see that the further imag of the series lie at distances which increase by equal step whereas this is not the case with the earliest images. If the equal steps are multiples of the wavelength, the contributio of the separate images will be in phase and reinforce each oth strongly, points in the room lying nearest to the image lin attaining the highest pressures.

The characteristic image-row frequencies are given by th formula

$$\tfrac{1}{2}c/\sqrt{[(n_1\alpha)^2 + (n_2\beta)^2 + (n_3\gamma)^2]} \quad . \quad . \quad . \quad (4$$

re α, β and γ are the dimensions of the room and n_1, n_2 and an be any whole numbers or zero.

s with the random images, however, the decay rate is a duct of r^{kt} and the inverse of the time. After an initial high plitude, therefore, the pressure due to these image-row systems apses quickly and they contribute little to the long-term rberation.

inally, we have the development of the true room modes ch are formed by the propagation of approximate plane waves n sets of images arranged in planes. To understand the in which these modes build up, we remember that each ge constitutes a point source of sound from which spherical es are propagated. If we consider the wavefront coming n a whole array of images lying in one plane, we see that at a rt distance from the plane the wavefront consists of a set of ial spheres intersecting at their edges. As we go further y from the image plane, these spherical surfaces approach rer and nearer to planes until at a great distance they coalesce orm a single plane wavefront. The distance, and hence the e, before this process can be regarded as effective is shown by yo to be much greater than that for the establishment of ge-row frequencies.

lowever, since the wavefronts are plane and do not diverge, e is no inverse-distance attenuation, and the rate of decay ends only on the reflection coefficients of the walls. Once blished, therefore, these true room-modes take a long time lie away, and constitute the main reverberant energy in the m.

he frequencies corresponding to these modes are determined the distance between the adjacent planes of images, and are esented by the formula

$$\frac{c}{2}\sqrt{[(n_1/\alpha)^2 + (n_2/\beta)^2 + (n_3/\gamma)^2]} \quad . \quad . \quad . \quad (5)$$

reference to Fig. 1 it will be seen that the planes with the itest numbers of images will be those running parallel to the s of the room. The modes to which they give rise are ally described as 'axial', and since only one dimension of the n is involved, two of the n's in expression (5) are zero and it ices to the simpler form of eqn. (1). Similarly, the tangential les have one zero n and oblique modes no zero n's.

he individual contributions of the higher harmonics of the ple modes are weak, and those of the high-frequency funda- itals (which are necessarily tangential or oblique) are quite igible because the images are widely spaced within the plane. Ve can now see the way in which the reverberant sound sure will rise and decay during and after the utterance of a le syllable of speech or note of music. At first there will be reinforcement of the direct sound by randomly spaced reflec- s reaching a high initial amplitude but decaying sharply after end of the syllable. The effect will be greatly dependent the positions of the source and microphone. Next, the ge-row frequencies will appear, with high steady intensity but pid decay. The true modes will meanwhile be building up vly, acquiring only moderate intensity, but decaying slowly ugh to provide the bulk of the audible reverberation.

he relative importance of the two series, having regard to r build-up and decay times and their maximum intensities, ends on the mean reflection coefficients of the walls. For the of a typical small studio with a mean absorption coefficient ibout 0·3, calculation shows that no frequency is likely to ome prominent unless it is common to both image-row and ge-plane systems, giving a high early intensity and a long iy. This condition is satisfied only by the axial modes, ch are therefore the only ones likely to become individually ificant. An exception to this rule is that a few tangential or oblique modes of low frequency may possibly be audible, owing to their high initial intensities or wide spacings.

The analysis of a room is therefore considerably simplified since the frequencies of the axial modes form three simple arithmetic series. Calculation of the tangential and oblique modal frequencies is much more laborious, for their total number increases roughly as the cube of the frequency. This is illustrated by Table 1 calculated for an experimental talks studio of typical size.

Table 1

NUMBERS OF AXIAL, TANGENTIAL AND OBLIQUE MODES OF A TYPICAL TALKS STUDIO

Frequency limits	Axial	Tangential	Oblique	Total
c/s				
0–50	2	0	0	2
50–100	3	6	1	10
100–150	4	11	7	22
150–200	3	19	16	38
200–250	3	19	29	51
250–300	4	30	51	85
300–350	2	34	69	105
350–400	3	46	98	147

(2.4) General Conditions for Audibility of Room Modes

Having established that the axial modes alone are likely to have sufficient amplitude and duration, we can now examine the other conditions for their audibility. A mode is heard as a 'coloration' if there is a noticeable tendency for reinforcement at the modal frequency or for sound of neighbouring frequencies to rise or fall in pitch towards that of the mode during the decay time. These processes will be expected to occur if there are, in the frequency region considered, strongly excited modes separated by a frequency interval great in comparison with their bandwidths. Now, it has been shown above that the axial modes form three series with uniform spacing throughout the audible frequency range and that the bandwidths are likely to be about the same for all members of a series. Therefore, we might expect the modes to be equally audible in all parts of the voice-frequency range; in fact, the region of audibility is restricted for other reasons. The most important of these is the presence of tangential and oblique modes, which, though virtually absent at frequencies for which the wavelength is comparable with the room dimensions, are far more numerous than the axial modes in the majority of the voice-frequency range. These non-axial modes, though not individually significant, dissipate an appre- ciable fraction of the sound energy, and reduce the intensity of the axial modes to such an extent that they are no longer prominent.

This was clearly brought out by some experiments on artificial reverberation[2] in which sound was prolonged by being repeatedly made to traverse a long tube. As the reverberation obtained from such a system exhibited strong resonances, the effect of having three tubes of different lengths to represent the three dimensions of a room was tried. This was exactly equivalent to a room having complete series of axial modes but no non- axial ones, and it was found that the results were very little better than those from a single tube, strongly excited modes being audible up to quite high frequencies.

The conclusion is, clearly, that the higher the frequency of a mode the less likely is it to be individually audible. For a good talks studio, modes above about 300 c/s are seldom distinguish- able; in living rooms without properly designed acoustic treatment, the limit is usually higher but in general not more than 1 000 c/s.

The above considerations, applying to the room as a whole, enable one to predict which modes are potentially audible. The two remaining factors listed in Section 2.1 must, however, be considered, since they affect the actual excitation of the modes. The position of the speaker or other source of sound in the room will affect the audibility, since a mode is most vigorously energized by a high-impedance source at a pressure antinode but not by a source at a node. Thus all modes will be equally excited by a source at a corner, but for most other positions there will be many modes that are only weakly excited. The expression 'at a corner' implies here that the source is less than a quarter-wavelength from the corner for sound of the highest frequency, a condition which cannot be realized in practice. A good alternative position, which ensures approximately equal excitation of all modes, is one-third of the way along a diagonal from one corner of the room to the corner farthest away from it. The same remarks apply reciprocally to the position of the microphone used in a broadcasting studio.

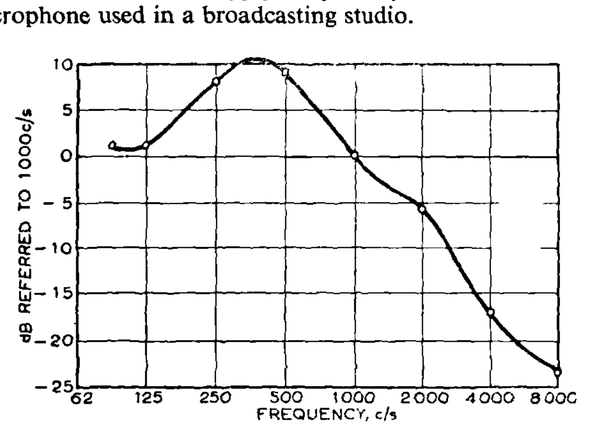


Fig. 2.—Energy distribution in male speech.

Ordinate shows speech power measured in ⅛ sec intervals (integrated over whole sphere) which is exceeded during 1% of whole time. Derived from data in References 23 and 24.

Lastly we must consider the energy spectrum of speech. The maximum energy, as shown in Fig. 2, is in the neighbourhood of 300–400 c/s. A more important feature, however, is the distribution of the fundamental and formant frequencies, which are considered in Section 3.

(3) COLORATIONS

(3.1) General Characteristics

The characteristic low-frequency colorations which take the form of an unnatural and often monotonous emphasis of certain frequencies in the speaker's voice are to most listeners the most objectionable feature of broadcast speech originating in a small studio, and much effort has been directed towards their elimination by acoustic design and treatment.

In general, a room will have many potential coloration frequencies, determined, as shown above, by the existence of prominent isolated modes. Only a few of these will actually be strongly excited by speech, however, owing to the fact that speech in its lower-frequency region is composed of a limited number of distinct pitches, the voice fundamentals and overtones. These vary from vowel to vowel, with changes in the inflection of the voice and with the individual speaker.

Purely subjective methods of assessing colorations will, therefore, tend to find a smaller number than objective methods, which should, if sufficient representative positions in the studio are examined, show all the modal frequencies of a type capable of giving identifiable colorations, including those of frequen not strongly excited by the human voice.

(3.2) Subjective Tests for Colorations

The method used by the B.B.C. Research Department is listen to several people speaking in turn at a microphone in studio, the voices being reproduced in another room by mea of a high-quality loudspeaker. The presence of any ov emphasized tones is noted, and estimates are made of the f quency and severity of each one. A most successful instrumen aid is a selective amplifier which is arranged to amplify a narr frequency band to a level about 25 dB above the rest of t spectrum. The output is fed in small proportions in paral with the original signal to the loudspeaker, the proportion bei adjusted until it is barely perceptible as a contribution to t whole output. Any colorations can then be heard clea when the selective amplifier is tuned to the appropriate frequen

This process is carried out as a routine when testing tal studios, to determine the frequencies of the most obvio colorations. In general, only one or two obvious coloratio are found in a studio and it is of interest to see how these a usually distributed.

Fig. 3 shows an analysis of 61 colorations observed in tal

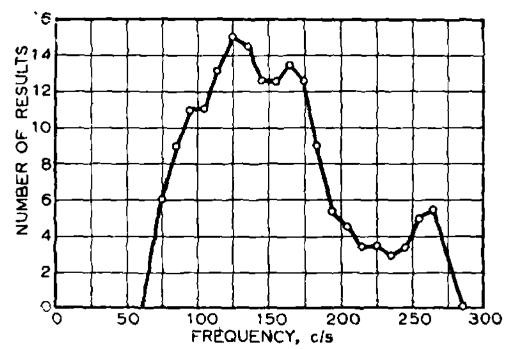


Fig. 3.—Frequency distribution of observed colorations in studios (61 observations).

studios during a period of about two years, using male speec as the programme material. The horizontal scale represents th centre values of frequency bands 10 c/s wide and the vertica scale shows the number of results falling into those bands. T avoid giving undue significance to a chance large number o results in a single column, each ordinate represents the runnin average with the two on either side of it. It is clear that mo colorations fall into the range 100–175 c/s and that there is subsidiary maximum at about 250 c/s. There is insufficien information on women's voices to plot a similar histogram, bu they almost invariably show the strongest colorations betwee 200 and 300 c/s. Obvious colorations below 80 c/s are ver rare, and those above 300 c/s become decreasingly prominent partly because of the diminution of speech energy and partl because of the increasing numbers of non-axial modes sharin the total energy above this frequency.

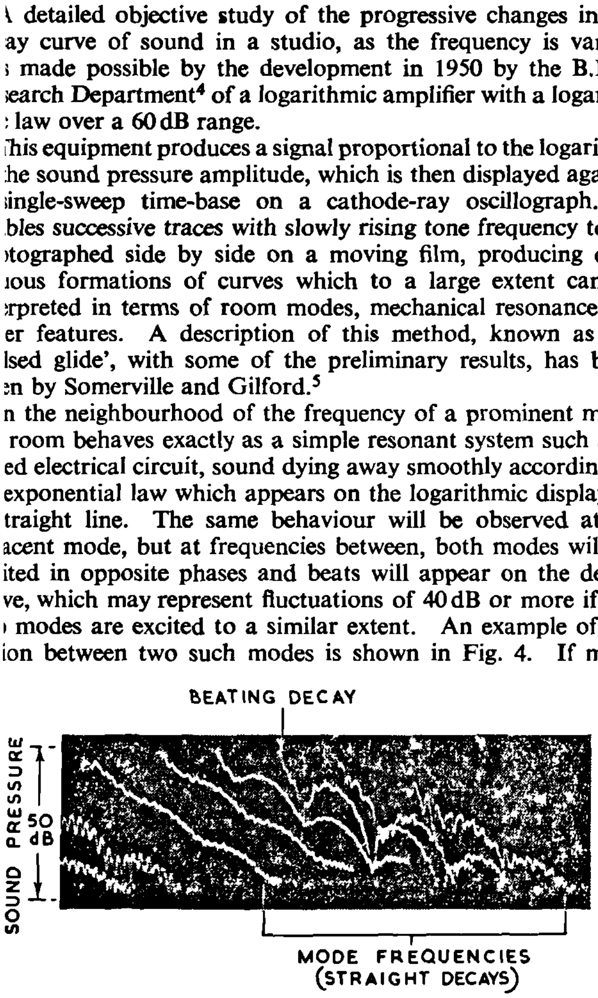
The experimental distributions shown in Fig. 3 should b compared with the known spectrum of the voice. Peterson and Barney[3] have measured the fundamental and vowel-forman frequencies for men, women and children. They found that on the average, men produced fundamental frequencies rangin from 124 to 141 c/s according to the vowel, but that there were large individual variations from this range, the standard devia tion being of the order of ±25 c/s. Women, on the other hand produced average fundamental frequencies ranging from 210 to 235 c/s according to the vowel. The first formants of male ech start at 270 c/s and those of female speech at 310 c/s, in with variations of a similar order according to the indiual. These figures, compared with the test results of Fig. 3, gest very strongly that the majority of audible colorations studios are those directly excited by either fundamental or t formant frequencies in the speaker's voice.

(3.3) Instrumental Detection of Potential Colorations

\ detailed objective study of the progressive changes in the ay curve of sound in a studio, as the frequency is varied, s made possible by the development in 1950 by the B.B.C. search Department[4] of a logarithmic amplifier with a logarith- : law over a 60 dB range.

This equipment produces a signal proportional to the logarithm he sound pressure amplitude, which is then displayed against single-sweep time-base on a cathode-ray oscillograph. It bles successive traces with slowly rising tone frequency to be otographed side by side on a moving film, producing con- ious formations of curves which to a large extent can be erpreted in terms of room modes, mechanical resonances or er features. A description of this method, known as the lsed glide', with some of the preliminary results, has been en by Somerville and Gilford.[5]

n the neighbourhood of the frequency of a prominent mode room behaves exactly as a simple resonant system such as a ed electrical circuit, sound dying away smoothly according to exponential law which appears on the logarithmic display as traight line. The same behaviour will be observed at an acent mode, but at frequencies between, both modes will be ited in opposite phases and beats will appear on the decay ve, which may represent fluctuations of 40 dB or more if the modes are excited to a similar extent. An example of the ion between two such modes is shown in Fig. 4. If more



4.—Pulsed-glide displays showing two adjacent modes with beats on intervening decays.

1 two modes are simultaneously excited, the beat pattern become more complex. Where there are many modes, as a large studio, or a small studio at high frequencies, the tuations from the exponential law become virtually random. he unfluctuating straight-line displays separated by regions with clearly defined beats are unmistakable indications of strong room modes. Even small rooms show only a small number of such formations, however, out of the hundreds of modes which exist. This verifies the conclusion reached above, that most room modes have little individual effect.

Another characteristic formation is caused by the presence in the room of a mechanically resonant object, such as undamped wall panelling or a metal radiator or lampshade. Such objects are usually forced slowly into oscillation because of high inertia and never become important sources of radiation. However, after the exciting sound has ceased, the vibrations die away more slowly than the room modes, becoming the chief source of sound for the later part of the audible decay. Fig. 5 is an example of a pulsed glide display showing this phenomenon.

To make a clear-cut distinction between these two types of display would be misleading, because wall resonances can give displays similar to those of room modes, and, conversely, an isolated room mode which has very low damping can show a second slope late in the decay curve.

Experience with these methods during the last six or seven years has shown that, provided reasonable precautions are taken to eliminate structural resonances, colorations are almost always associated with room modes. A defect of the pulsed-glide method is that the appearance of the display depends greatly on the position of the microphone, and to assess a room completely it is necessary to repeat the glide at several different microphone positions, deducing the importance of the several features by an inspection of all the displays. A development of the method was therefore tried in which the pressure amplitude display was replaced by one showing the scalar product of the sound pressure and the oscillator signal which produced it. A full account of this test and the results obtained with it were given in a recent B.B.C. Engineering Monograph.[6] The displays, of which Fig. 6 is an example, show characteristic fluctuations, the number of which represents the change of frequency undergone by the sound in its transformation into reverberant energy at a modal frequency. Since the recognizable features of the displays are mainly determined by the frequency information, they are largely independent of the amplitude and consequently are less sensitive to the degree of excitation of a particular mode at each microphone position. It is also probable that the audibility of a coloration is partly connected with the changes of pitch of programme material of neighbouring frequencies, and hence that the test corresponds closely to sensation. It gives rather more reliable predictions of colorations than the simple amplitude display, and is able to resolve neighbouring modes only a few cycles apart. The method has been used more as a laboratory technique than a routine test, which presents difficulties in practice.

(3.4) Design Precautions for Avoidance of Colorations

Having now reviewed the influence of mechanical resonances and of room modes of different types and frequencies, we are in a position to decide on the correct design procedure. Taking

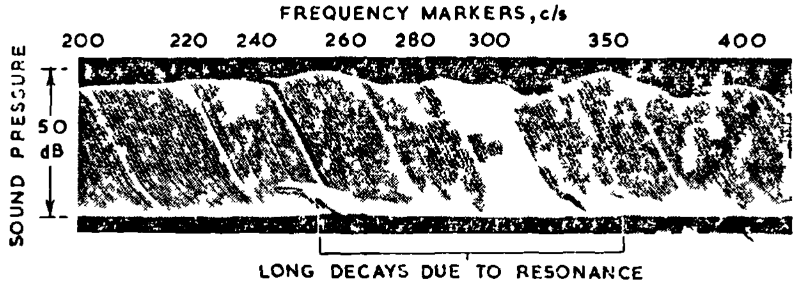


Fig. 5.—Pulsed-glide display showing mechanical resonances.

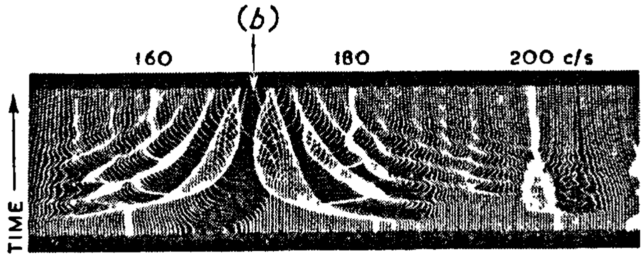


Fig. 6.—Part of coherent glide, showing strong mode at (*b*).

Frequency markers are shown on the top margin, and the horizontal displacement of each trace from zero (approximately at its intersection with the top margin) represents the scalar product of the input tone and output microphone signals.

structural resonances first, little need be said except that structural and lining materials having high Q-factors should be avoided. The materials with which effects of this sort have most frequently been associated in the past are plaster on expanded-metal lathing, plywood, breeze block, and plasterboard with an unsupported back surface.

The problem of room modes is fundamentally more difficult, because they cannot be eliminated or indefinitely reduced without at the same time eliminating all or most of the reverberant sound. There are objections to this, as will be explained below, and we are therefore faced with the more difficult problem of making the unavoidable modes less conspicuous.

Attention must first be given to the dimensions of the room. A simple calculation based on eqn. (1) enables a list of all the axial modes for all three dimensions to be written down in order of frequency. It will be unnecessary to continue the list beyond, say, 350 c/s because, as already noted, the axial modes in a well-designed talks studio will not be prominent above that frequency. The list must next be examined to find modes, or groups of modes with almost the same frequency, which are separated from their nearest neighbours on either side by intervals appreciably larger than their bandwidths. In practice the minimum separation for audibility appears to be about 20 c/s.

Modes or groups separated from their neighbours by greater intervals than this should be noted, and if they fall in the frequency ranges likely to be excited by voice fundamentals and formants attempts should be made to alter the groupings by changes in the proposed room dimensions. It is impossible not to have some isolated groups within the list, but it is usually possible to avoid very bad examples.

Reducing the reverberation time of the room at a particular frequency increases the bandwidth of the modes and reduces their excitation. The application of selective absorption at the frequency of remaining isolated groups would therefore be a possible method of controlling them; this has been tried in the past, sometimes with success, but a limit along these lines is set by the fact that very selective absorbers such as may be required have themselves long decay times and a tendency to reradiate absorbed sound.

However, it is often useful to apply a selective absorber on the walls perpendicular to the longest dimension of the room, because for a given absorption coefficient the axial modes for this dimension will have the smallest bandwidth and will therefore be most likely to be audible.

(3.5) Distribution of Absorption Coefficient between the Boundaries

~~So far, no account has been taken of the influence of differences~~ in absorption coefficient between one pair of walls and another. In Section 2, it was assumed throughout that all the surfaces had approximately equal average absorption coefficients, and the calculations of the relative importance of the different types mode were based on this assumption. If, however, all t absorbers are concentrated on two pairs of parallel surfaces, a the third pair is substantially reflecting, strong axial modes v be formed between the latter, all other modes of the room bei suppressed. The listener will then hear only one harmonic ser of modes, which will remain separate and distinct up to very hi frequencies. These conditions give rise to the well-kno phenomenon of flutter echo, any time function of the pressu at a source being reproduced periodically with a time inter determined by the distance between the two surfaces in questic It should be emphasized that this can occur only when one p of surfaces is very much more reflecting than the other two pai It is an effect of relative rather than absolute reflection coefficie and does not occur, for example, in a tiled reverberation roo where all the surfaces are highly reflecting. The most famili example is that of the space between two high walls.

If two pairs of surfaces in a room are highly reflecting a the third pair absorbent, there will be two harmonic series axial modes, some tangential ones but no oblique ones. T audible modes will extend to considerably higher frequenci than in the case of a uniform room, but the highest-frequen modes will be indistinguishable and there will be no flutter-ec formation. Instead, impulsive sounds will excite a series clear musical rings up to frequencies of the order of 1 000 c/s.

These conclusions on the behaviour of non-uniform roo were verified by experiments during the course of the constru tion of some studios in Portland Place, London. By successi addition of absorbing material it was established that flutte and rings were likely to be noticeable features of the acousti of any room if the ratio between the mean absorption coefficien of any two pairs of walls was greater than about 1·4 : 1. designing studios of small or moderate dimensions this rat should not in any circumstances be exceeded at any frequenc and the aim should be for ratios nearer to unity, with slight higher mean coefficients for the pair of walls with the greate separation. The reason for this reservation was given Section 3.4.

(4) DIFFUSION

Much has been written on the influence of diffusion on t acoustics of studios. A sound field is said to be diffuse if any moment the intensity of the sound is uniform over the who volume and if at any point the energy flow is the same in a directions. The implications are that there are no predomina standing-wave systems and that no one position can be di tinguished from another. This is an unattainable ideal, but sound-field may approach it more or less closely, being said have a greater or less degree of diffusion. The degree diffusion may be measured by measuring the statistical variatic of certain acoustic properties with position, direction, frequen or time. The most satisfactory test, in the author's experienc is to examine the departure of typical decay curves from generally straight course. Fig. 7 shows two sets of dec curves, (*a*) being from a room with a fairly diffuse sound fie and (*b*) from one in which the average damping for modes one direction was much greater than that for the other direction In the latter case, the most highly damped modes vanish firs leaving the less damped modes to determine the slope of the lat part of the decay.

Many authors have laid down rules for the improvement ~~studio acoustics by increasing the degree of diffusion. It~~ ~~usual in many organizations to avoid parallel walls, to~~ introdu irregularities in the wall surfaces and to distribute the absorbin materials as a series of irregular areas over the walls and ceilin The subjective evidence for the advantages of any of the

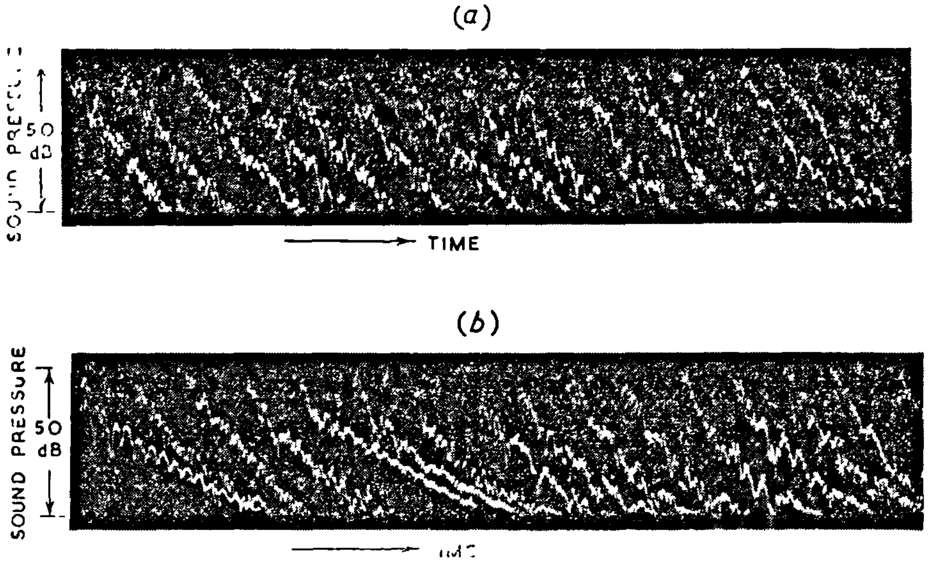


Fig. 7.—Decay curves from diffuse and non-diffuse rooms.
(a) Diffuse room: slopes of curves are substantially constant during decays.
(b) Non-diffuse room: slopes diminish as decay proceeds.
Each curve represents a different frequency in the range 150–250 c/s.

sures has for the most part been inconclusive, almost all fication having been by reference to theoretical considera- s or the results of purely instrumental measurements on the istic properties of the rooms in question.

'e will therefore review the evidence in favour of these sures, remembering that the final justification must always n agreed improvement in quality.

(4.1) Effect of Wall Angle

common method of increasing diffusion is to build the osite pairs of walls a few degrees out of parallel. This is lly said to eliminate flutter echoes, but in practice does not so entirely. It has been reported that a room with non- llel walls gives a more regular frequency characteristic than ctangular one. Nimura and Shibiyama,[7] working in model ns of which the shapes could be varied, have measured the uency irregularity[8] which is a measure of the extent to which steady-state level in the room due to a source of constant igth fluctuates with frequency. They conclude that non- llel walls reduce the irregularity below a frequency of about :/s, the maximum effect being produced by an angle of 5° een opposite walls. Schröder[9] has shown theoretically and rimentally that the frequency irregularity is directly propor- al to the reverberation time, and that changes which affect reverberation time will therefore also affect the irregularity. , the absorption coefficient of a surface to grazing-incidence id is generally half that to normally-incident sound. This been explained by the author in connection with resonant rbers.[10] Non-parallel walls, by discouraging the establish- t of repeated grazing reflections, will tend to improve the iency with which the absorbers act, reducing the reverbera- time and hence the irregularity.

must be borne in mind also that the models used for these riments had uniform flat walls, whereas this is not normally ase in a studio. It therefore remains to be seen whether the lts are valid under practical conditions.

ill-scale experiments were carried out by the B.B.C. in the part of this decade, to determine whether differences of angle alone were subjectively significant. An experimental studio was built in which the angles of two of the walls could be varied up to a maximum of 6°. Recordings of male and female speech made in parallel and non-parallel configurations were then compared by panels of experienced listeners, but there was found to be no concordant preference for one condition or the other.

It should be remarked here that the wall surfaces were treated by irregular patches of absorbers to give the desired reverbera- tion/frequency characteristic. The conditions were therefore not those of uniform wall surfaces always assumed in the theoretical and experimental evaluation of wall angles. A full-scale experiment avoiding this divergence from 'ideal' con- ditions is impracticable, since a single uniform absorber with the absorption/frequency characteristic required does not exist.

We will therefore consider now the effects of the perturbation of the individual wall surfaces and the subdivision of absorbing materials as a means of increasing the degree of diffusion in a room, and hence of ameliorating the effects of strong modes on speech quality.

(4.2) Effect of Wall Irregularities

The first work published since the war was that of Somerville and Ward,[11] who showed by means of small-scale models that the perturbation of walls by projections had a measurable effect on the diffusion for sound of wavelengths less than about one- seventh of the height of the projections. Rectangular prisms were found to be more effective than triangular prisms or hemi- cylinders of the same volume, and this result was also demon- strated theoretically by Head.[12]

These conclusions were at variance with the then current opinion that cylindrical diffusers were the most efficient. Meyer and Bohn,[13] for example, found that in free-field conditions hemicylinders produced the greatest scattering effect on plane waves, but free-field results are not necessarily valid in relation to the standing-wave field in a room. The work of Somerville and Ward was later confirmed experimentally by Bruel,[14] and experience in concert halls and large music studios tends to lead to the same conclusions.[15]

However, although in large spaces such as these the breaking up of wall surfaces by projections is both effective and necessary, it has disadvantages when applied to small studios, since to be effective the projections must be bulky. The wavelength at the commonest coloration frequency (see Fig. 3) is about $2\frac{1}{2}$ m, and to have a worth-while effect the depth of the projections would have to be appreciably greater than one-seventh of this amount, say 50 cm. Not only would such projections be highly inconvenient in a small studio, but the relatively deep recesses formed between them would exhibit their own resonances, a phenomenon often observed with similar accidental features in rooms.

Thus the shallow, irregular, polycylindrical diffusers so often used in studios abroad can hardly be expected to have any effect at all on the most serious bass colorations, though they may have other subjective effects associated with higher frequencies. A few attempts to establish such an effect by subjective tests have been made.

Jeffress, Lane and Seay[16] used two rooms with identical dimensions, one having plane walls and the other polycylindrical. Both were devoid of added absorbing material and both had identical reverberation characteristics. In spite of the difference in wall configuration, however, comparable word-intelligibility tests gave substantially equal mean results of 75·4 and 73·9 respectively.

Within the last few months, a comparison of speech quality between flat and coffered walls has been made in two studios in the B.B.C.'s new centre in Bush House, London. These two studios were designed with the intention of making such a direct comparison possible, and they were therefore arranged to be identical in dimensions, structure, reverberation time and distribution of the absorbing materials. The reverberation characteristics of the two studios are within 0·05 sec of each other at all frequencies. The results of several series of listening tests showed that the audible difference between them was very small—actually less than the variation with microphone position in either studio.

(4.3) Effect of the Irregular Distribution of Absorbing Materials

The other known method of introducing diffusion by scattering is to make the surfaces non-uniform with respect to absorption coefficient, the necessary absorbers being distributed in comparatively small areas. The diffraction effects at the edges of these areas cause the sound to be scattered over a wide angle, thus breaking up the standing-wave patterns. The absorbers are normally much shallower than the diffusers described above, and therefore less wasteful of room spaces. It has been stated, however, that this method is less effective than altering the surface shape,[17] and the B.B.C. Research Department therefore undertook an investigation of the relative effectiveness of the two methods, with full-scale experimental material.

These experiments were conducted in a tiled reverberation room of $27\,m^3$ volume, and were confined to frequencies above 500 c/s, for which diffusers of reasonable depth would be adequate and room modes not isolated. The criterion for the diffusion was taken as the apparent absorption of an area of absorber entirely covering one wall of the room. With non-scattering walls, the absorption coefficient measured by the reverberation method is low because the decay curves obtained are dominated in the later stages by the modes that avoid incidence at near-normal angles on the absorber.

One wall of a reverberation room was covered entirely with an absorbing material and the total absorption was calculated from measurements of the reverberation time. This was repeated with the same material distributed as five patches on different room surfaces. The latter condition gave a result about 75% higher than when the whole material was on one surface. The whole experiment was repeated, first with twelve stout woo[den] boxes hung on the walls, and then with the boxes replaced [by] patches of efficient absorbing material of equal total area. [In] both cases the addition of the diffusing elements, whether bo[xes] or absorbers, had very little effect on the absorption of the [dis]tributed absorber, but increased that of the single-wall arran[ge]ment to a figure comparable with that given by the distribu[ted] material. The conclusion from these experiments was that, [by] the criterion adopted, rectangular irregularities and patches [of] absorber of similar size were equally effective as diffusers wi[thin] their specific frequency ranges of action. Low-frequency abs[or]bers of the membrane or Helmholtz resonator type are, a[s a] rule, very much shallower than the depth recommended ab[ove] for projections. They are therefore clearly preferable to p[ro]jections when good diffusion down to the lowest voice f[re]quencies is required in small studios.

Summarizing the conclusions of this Section, it appears t[hat] in small studios no advantage is to be gained from the use [of] non-parallel walls or diffusing projections, provided that [the] simpler and more convenient expedient is followed by [dis]tributing the absorbers in small areas over as many surfaces [as] possible.

(5) WHAT MAKES NATURAL SPEECH?

(5.1) The Monaural Chain

Up to the present we have been concerned almost exclusi[vely] with the standing-wave systems in small rooms and with [the] colorations that arise from them. There are, however, ot[her] matters of which we must take account in the quest for go[od] speech quality. The basic problem in broadcasting un[der] existing conditions is that there is only one channel betw[een] the studio and the listener, whereas we are accustomed to h[ear] speech and music by means of two ears which together sup[ply] information about the direction and position of the source.

When listening directly with two ears one is provided with [an] automatic mechanism for partially rejecting sound other t[han] that coming from the direction of the source to which one [is] listening. This is an evolutionary faculty possessed by all [the] higher animals; when it is inhibited by having only one chan[nel] of information one is conscious of the reverberant sound a[nd] extraneous noise to such an extent that speech loses its intell[igi]bility and music its definition unless steps are taken to incre[ase] the ratio of direct to reverberant sound.

(5.2) Reverberation Time

The subjective balance between the direct sound and unwan[ted] sound made up of reverberation and extraneous noise may [be] restored most easily by reducing both these components. T[he] reduction of extraneous noise is a matter of providing effect[ive] sound insulation for the studio and giving attention to the no[ise] generated by ventilation systems and other sources inside t[he] room.

The reverberant sound is reduced simply by the applicati[on] of sound absorbers. A certain amount of low-frequen[cy] absorption is provided by the compliance of the room structu[re] itself, sound energy being dissipated by frictional losses as t[he] walls, floor and ceiling vibrate. Unfortunately, good sou[nd] insulation in practice necessitates building massive walls of l[ow] compliance, and in these circumstances there is no gratuito[us] absorption, such as is derived from the floors, ceilings a[nd] windows of ordinary houses.

It is therefore necessary to add considerable extra absorpti[on] to reduce the reverberation time to a point where speech by t[he] monaural listening chain has the intimacy of conversation in [a] well-furnished room.

One important effect of reverberation should be noted here. the studio is very non-reverberant or 'dead', the resulting ech comes from the listener's loudspeaker substantially changed by any added reflections from the studio walls. It then modified by the reverberation in the listener's own room exactly the same way as the voice of an occupant, and the ision of actual presence is created. Conversely, reverberation ded in the studio will be recognizable as foreign to the room d will give the illusion that the loudspeaker is a hole in the ll communicating with another room where the broadcaster sitting. Different individuals have definite preferences for one other of these conditions, and, in the author's own experience, ost of those engaged in the engineering or production aspects broadcasting in this country prefer the illusion of presence. should be remarked, however, that Continental broadcasting ganizations seem to prefer a very much more reverberant und.

Whether the individual's preference could be correlated with general psychological make-up is an interesting speculation tside the competence of an engineer. The broadcaster himf will, however, disagree with the implications of the majority oice, because to speak in a very dead studio is unpleasant, oping the confidence of all but the most experienced news ader and compelling the speaker to raise his voice in an effort obtain the reassurance given by the reflections which would nforce the voice in other circumstances.

The type of microphone is important in this connection. An nidirectional microphone, such as a moving-coil or piezoctric instrument, increases the ratio of reverberant to direct und, thus requiring a deader studio and increasing the strain the broadcaster. Ribbon microphones with a figure-of-eight aracteristic are rather better, since they reject part of the verberant sound, and the directional characteristics may be ed to discriminate against the strongest modes. Normally, ere the diffusion in the studio is fairly good, a ribbon microone placed diagonally across the room is usually found to be st since it reduces axial modes from the two horizontal direcns by about 3 dB with respect to the rest, whilst discriminating o against up-and-down modes. A cardioid microphone duces the apparent liveness of the studio but does not much er the relative effects of different modes.

(5.3) Shape of the Reverberation Characteristic

The quality of speech is critically dependent on the shape of reverberation-time/frequency characteristic. Past experience d controlled experiments have combined to show that the verberation time should be independent of frequency from c/s to 8 kc/s. Deviations from a level characteristic are easy recognize; excess of low-frequency reverberation produces oominess' and distinct colorations, while excess in the region 0–500 c/s is most unpleasant, giving a throaty, strangled ality to speech. A long reverberation time (above 2 000 c/s) oduces sibilance or 'breathiness'. A slightly drooping characistic below 500 c/s is perhaps ideal, giving the most natural eech quality.

(5.4) Microphone Correction Circuits

It is a common device to include in the microphone circuit a er to give slight bass attenuation. This is usually carried out trial until acceptable speech, free from boominess, is obtained. orrection may be required in some cases to compensate for a e in the reverberation time below, say, 100 c/s, but there is other reason. Most people listen to broadcast speech at a her level than that of the broadcaster's voice in the studio or a friend speaking to them in person. Somerville and ownless[18] found that members of the public preferred to listen to an average level of 71 dB above the standard reference level of 10^{-16} watt/cm^2, whereas the typical level of conversational speech is about 60–65 dB. A well-known property of the ear[19, 20] is that the equal-loudness contours plotted against frequency approach each other closely at low-frequencies. Speech will therefore sound bass-heavy if it is reproduced at an unnaturally high level. A bass cut of 3 dB at 50 c/s relative to 250 c/s would be of the right order to correct for the difference between average listening levels for broadcast speech and live conversation. A pressure-gradient microphone closer than about 60 cm also requires a cut of a similar amount to correct for the curvature of the wavefronts. Compensation for long reverberation in the bass by further electrical equalization is possible to a limited extent only, since the effects of colorations, being frequency-selective, can only be removed by cuts of such a magnitude that an emasculated quality is imparted to the speech.

(5.5) Influence of the Listening Room

The listener's own room has an influence on the transmitted speech since it adds colorations and other effects of reverberation in the same way as the studio. However, although any additions from this cause are reduced by the binaural rejection mechanism and assume relatively less importance, they are an impediment to critical listening and will therefore be considered in Section 7.3.

(6) THE DESIGN OF TALKS STUDIOS

(6.1) Size and Shape of Studios

The dimensions of a talks studio should be large enough to give reasonably close spacing to the axial modes. Volumes from 1 500 to 4 000 ft^3 (43–114 m^3) are generally satisfactory, bad colorations being difficult to avoid in studios below this range, and larger studios giving insufficient advantage to justify the increased expense of construction and treatment.

It is doubtful whether any of the preferred ratios between dimensions, published from time to time, can be upheld. Certainly the once popular 5 : 3 : 2 ratios will normally give at least one isolated group within the worst frequency range. Neither can one depend on proposed ratios based on a consideration of modal frequency-spacing statistics,[21] since the presence of a single isolated group, which may not greatly affect the mean frequency spacing, will give a serious coloration. The only way that has had any success, in the author's experience, is to work out the axial-mode frequencies for a set of trial dimensions, as described in Section 3.4, and to adjust the dimensions until a satisfactory mode spacing is achieved. Tests of these several theories were made in an experimental studio in the B.B.C. Research Department in which one of the walls, constructed of heavily reinforced clinker block, could be moved perpendicularly to its plane to give dimension ratios predicted by various theories as 'good' or 'bad'. The results confirmed substantially the over-riding importance of isolated axial modes, but failed to show any difference between configurations with high or low values of the frequency-spacing statistic.

The shape of the studio may be rectangular, or the walls may be splayed at angles up to a few degrees if the shape of the site renders this more convenient. There is insufficient evidence for a dogmatic view about these questions, and one cannot rule out completely non-rectangular shapes such as triangular or pentagonal prisms, which have been used by other organizations. There have been suggestions, however, that by their unfamiliarity in ordinary homes, such shapes impart a somewhat unnatural quality to speech. Irregularities in the wall shapes are, as shown in Section 4.4, unnecessary.

(6.2) Reverberation Time

The reasons for the choice of reverberation time have been dealt with in Section 5.2. B.B.C. experience has shown a time of about 0·3 sec to be the optimum, with a possible slight reduction below 300 c/s. The additional bass absorption required to give this reduction is difficult to achieve without making full use of structural absorption from the walls, floor and ceiling of the studio. Extra absorption is obtained by the use of resonant membrane absorbers consisting of sheets of bituminous roofing felt sealing an air space.[10] Helmholtz resonator absorbers have also been used, since they have the advantage that they can be tuned to the required frequency and adjusted for bandwidth and maximum absorption by the addition of resistive materials in the necks.[22]

Continental talks studios, as pointed out also in Section 5.2, tend to be more reverberant, and times up to 0·5 sec or even more are encountered.

(6.3) Application of Absorbing Materials

The aim, as described in Section 3.5, is to distribute the low-, middle- and high-frequency absorbers equally on each of the pairs of parallel wall surfaces, except that low-frequency absorbers should be slightly in excess on the end walls, i.e. those separated by the longest distance. On the individual surfaces there should be patches of efficient absorber surrounded by areas of poor absorber such as hard plaster. The absorbers should be arranged on the individual pairs in such a manner that there are no large areas of reflecting surface directly facing each other, since these will give rise to flutters.

The effectiveness of any absorber depends markedly on the position it occupies in the room. Calculation of the reverberation characteristic of a small room is therefore never very precise, and provision must be made for adjustment of the absorbers after completion of the studio. It is the usual practice in the B.B.C. to construct all absorbers, wherever possible, with detachable covers which can be removed for alteration, replacement or removal of the contents. The cover itself may act as a filter or the inductive element of a Helmholtz resonator, in which case it is made of a perforated board. Alternatively, it may be of fabric. Perforated covers tend to limit the performance of the absorber at high frequencies, thereby producing excessive sibilance, while fabrics soon become dirty and may shrink while being cleaned so that they cannot be refitted. This disadvantage has been overcome in a recent B.B.C. design in which the fabric is stretched over a detachable frame arranged to accommodate surplus fabric to compensate for shrinkage.

(7) DESIGN OF CONTROL CUBICLES AND OTHER LISTENING ROOMS

(7.1) Design Principles

It has already been said that, from the point of view of acoustic faults and the methods of overcoming them, listening rooms may be treated in the same way as small studios. Their different function, however, imposes differences in design, which will be briefly considered in this Section.

(7.2) Resemblance to Living-Room Conditions

Unlike electronic and electro-acoustic equipment, for which one can set an ideal input/output characteristic as a goal for both broadcasting organizations and their listeners, a domestic living room must be accepted very much as it exists. Control-cubicle and quality-monitoring-room acoustics should therefore not be very dissimilar from the average conditions encountered in private houses. A few years ago an acoustic survey of living rooms was carried out, embracing various types of constructi from Georgian houses in Pimlico to post-war maisonettes w solid floors in the suburbs. Fig. 8 shows the mean reverbe

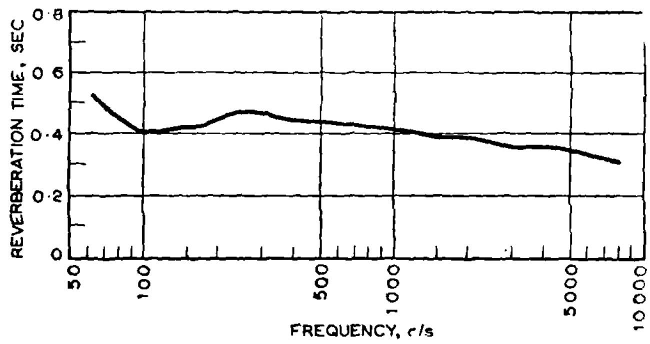


Fig. 8.—Mean reverberation characteristic of 16 living-rooms.

tion characteristic for all the rooms surveyed. There w surprisingly little variation in the results for conventional furnished rooms, though the reverberation times of rooms wi joist floors averaged about 0·05 sec less than those of t immediate post-war houses with solid floors built directly on concrete foundations. Listening tests with recordings of spee from several studios played into the rooms showed that most them allowed the characteristics of speech from the differe studios to be distinguished, but some of the rooms introduce very severe colorations which entirely masked other effects. was decided to adopt the mean curve of Fig. 8 as a pattern f listening rooms, pending the completion of subjective tests c the effects of listening-room acoustics on the judgment of speec quality.

(7.3) The Effects of Listening-Room Acoustics on Judgment of Quality

The fact that the listening room does not have a predomina effect on quality is very largely due to the binaural mechanism This is particularly true of rooms used for listening to musi programmes, since, objectively speaking, the standing-wave effect in the room can accentuate particular notes by as much as 8 o 10 dB relative to their neighbours. Fig. 9 shows two pulse glides obtained by radiating the test tone into a studio, trans mitting it by the studio microphone and the listening-roon loudspeaker and photographing the resulting decay curves a received by a microphone in the listener's position. The tw glides were obtained in the same listening room, but Fig. 9(*a* was from an anechoic sound-measurement room and Fig. 9(*b* was from a normally treated talks studio. It will be seen tha the glides are sufficiently similar to suggest that it was th listening room which had the principal effect on the overal acoustics.

Accordingly, since it was common experience, verified by the living-room tests described above, that studio differences were nevertheless distinguishable, subjective tests were undertaken to determine to what extent the binaural mechanism enabled one to reject the cubicle acoustics.

Recordings from six studios were played in four different listening rooms to panels of engineers. The test programme consisted of short passages read partly from one of the six studios and partly from another, and the subjects were required to state a preference for one part of the passage or the other Each studio was compared with every other in this way, making fifteen paired comparisons in all. Statistical analysis of the answers revealed the following information:

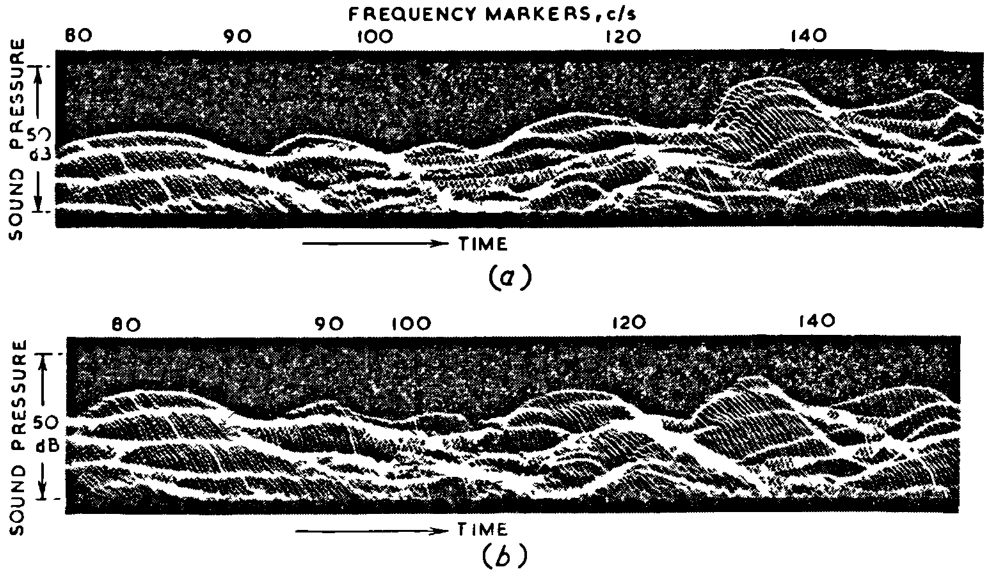


Fig. 9.—Combined pulsed glide of studio and listening room.

(a) Normal talks studio and listening room.
(b) Non-reverberant room and listening room.

ı) The average order of preference of the six studios.
ı) Differences in the rank order produced by using different ıning rooms.
:) The self-consistency of the individual subjects.
l) The effect of the listening room on self-consistency.

'he results showed that an over-reverberant listening room, le not altering the order of average preference, greatly eased the inconsistency of the subject's answers. A room ı a long reverberation time in the bass favoured a studio with ıavy bass cut, while a very dead room favoured a studio with g bass reverberation, presumably because a listener sitting roximately on the axis of the loudspeaker (the usual position critical listening) received an excess of high-frequency sound. 'he longest reverberation time which can be permitted without adverse effect on either the order of preference or the conency of judgment was approximately 0·4 sec, and it is to be ed that the average curve of Fig. 8 does not rise far above value.

ʌs the result of these investigations, all control and listening ms are now designed to have reverberation times of 0·4 sec to 1 000 c/s, falling steadily above this frequency to 0·3 sec :000 c/s.

t goes without saying that the same care must be exercised ıvoid serious colorations in listening rooms as is necessary he case of studios.

(8) CONCLUSIONS

'he acoustic design criteria for small studios and listening ms are given in the last two Sections of the paper. The nary requirement is to avoid the occurrence of axial modes arated by frequency intervals large compared with their ıdwidths. Such isolated modes are generally audible as ɔrations if they occur in the neighbourhood of the fundaıtal and formant frequencies of speech. It is impossible to ninate them entirely from small rooms without reducing erberation to an undesirable extent, but they can be eliorated by correct design.

'lutters and rings at higher frequencies are usually due to marked differences in the absorption coefficients between one pair of opposite boundary surfaces and the other two pairs.

Good diffusion of the sound field is accepted as necessary, and is best achieved by arranging the absorbing materials in irregular patches on the individual surfaces. The addition of diffusers as commonly understood is then unnecessary.

In listening rooms and control cubicles, the binaural listening conditions allow a longer reverberation time than in studios, though excessive reverberation makes critical listening more difficult.

(9) ACKNOWLEDGMENTS

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DISCUSSION BEFORE THE RADIO AND TELECOMMUNICATION SECTION, 15TH DECEMBER, 19

Mr. J. Moir: It is a major difficulty with problems of room acoustics that the final judgment of the room performance is almost entirely subjective. We know a great deal about the objective performance of a room, but we do not know what the objective performance should be in order to give an excellent result when subjectively judged.

Most of the paper is concerned with the acoustic design of talks studios and listening rooms. It might be qualified by saying 'listening rooms for talks', for there is very little comment about the conditions required for the reproduction of music.

Room modes are undoubtedly the main troubles in a small room, and the paper is particularly valuable in indicating that the axial modes represent the major difficulty.

I doubt whether the reverberation time at low frequency has much meaning. When we measure the reverberation time of a room, particularly in the region below about 150c/s (depending on the room size), the results generally vary radically with changes of frequency of only a few cycles per second. Curves commonly presented show a smoothed shape for the reverberation-time/frequency relation, but they are not so smooth in practice. It is not unusual to find that the reverberation time changes by a factor of 2 : 1 over a frequency range of a few cycles per second, and it is often doubtful whether the high or the low value is the effective one.

There has been a radical change of thought in recent years over the best shape of reverberation-time/frequency curve. At one time the optimum was assumed to be something rising fairly sharply at the low-frequency end of the range. In recent years, particularly as a result of experience with the Royal Festival Hall, there has been a tendency to favour a curve which falls off a little at the low-frequency end.

It has been claimed—and I think that measurements support it—that the majority of the old concert halls had reverberation-time/frequency curves which rose sharply at low frequency. I do not like music rooms in which the reverberation time falls off at low frequency, for the music appears to lack 'body' and roundness. There is no doubt that for the reproduction of speech, where intelligibility is the main factor, a reverberation-time/frequency curve which falls off at low frequency always gives better results than one which rises at low frequency; but music is often just as important as speech, and one may have compromise.

Flutter echoes have also been mentioned. They are v prevalent in cinemas. A hand clap near one wall is follow by a series of 20 or more 'slaps', and one is left with the impr sion that the acoustic conditions are poor, but we have ne found any real trouble from these flutters. I would like to h the author's comments on how troublesome they are in practi and indeed if they are a serious problem in studios.

Ringing at high frequency is much more serious. There very few living rooms in which there is not some evidence this. In my experience, non-parallel walls in small or lar rooms seem to produce a greater reduction in flutter echo a ringing than the author seems to have found. Non-paralleli of 5–10° appears to remove most of the flutter echo and ringi

Finally, in my view, if a room requires extensive treatment f stereophonic listening there is something wrong with the stere phonic equipment or the recording. The better the stereopho reproduction system, the less trouble we have with roo acoustics.

Mr. H. R. Humphreys: I am rather concerned by the gene inference that there is *one* good design for a talks studio. T reproduction of speech must be regarded as an artifice, sinc however natural the speaker may sound, he is not, in fact, the I think that this is justification for some variation in stud acoustics.

The author suggests a mean reverberation time of 0·3sec, b is this the right policy? Let us consider the different uses which talks studios are put. There is the single speake perhaps an announcer, and one might regard this as unaccon panied voice. There is discussion—duet, trio, quartet, et Then there is the commentator, which one might regard as concerto for voice and accompaniment.

A single speaker's voice may perhaps be allowed to issue the receiving end as 'disembodied' speech, but if this is do for a group of speakers in discussion we lose all sense of perspe tive. Only by using a limited and carefully controlled amou of studio acoustics can one get some sense of perspective on monaural channel.

The commentator will most likely be accompanied from tim ime by some 'noises off' or will be dubbed over a musical ound-effects background which will itself set the acoustical ıe. The major requirement for the voice is then intelligibility, :h can probably best be achieved against the distraction of ‹ground by using a fairly 'dead' or completely 'dead' studio. /ith regard to 'loud random reflections', a few weeks ago Beranek spoke on concert-hall acoustics, and he made a ıt about first reflections in these rooms. Does the author :ve that the first reflections in talks studios have any impor-:e, bearing in mind that the time intervals will be very much ·ter because of their comparatively small size?

he author states that, in some cases, structural resonances :aused by certain surfaces having a longer reverberation time ı the room itself, and he mentions 'plasterboard with an ıpported back surface'. I do not know what he means by because I cannot imagine using plasterboard without ›orting its back surface in some way or other.

ı rooms with non-parallel walls there is a tendency for the :tiveness of absorbents to be slightly increased. How much a absorption should one allow in designing for reverberation : when the walls are non-parallel?

Ir. C. G. Mayo: A studio, like a violin or loudspeaker, can ' respond in its own eigentones. Rather than being critical, n therefore surprised how well the studio reproduces the ıds emitted. If a note is sounded at a frequency between e of two eigentones reasonably close together, the response have the pitch of the note sounded, and not that of either ıtone. The pitch of the response is attained by a kind of frequency having a rapidly changing envelope. The zero sings give the right pitch, but the frequency is not that ciated with the zero crossings.

room having a very large number of eigentones or modes likewise respond to any frequency. If the modes were ılly spaced in frequency (as in an open-circuited cable) the ıd emitted would be precisely reproduced, but with a time y or echo. When eigentones are closely and randomly :ed, however, the input is faithfully reproduced. Thus a in or loudspeaker gives a remarkably good response in spite s limited power.

ı remote ages, the ear was used primarily for the rough tion of position and the determination of the size of an osure like a cave. The ear instantly estimates critically the of a talks studio or an auditorium.

is the absence of modes rather than their presence that spoils nall talks studio. A prominent single mode is annoying .use it lacks the necessary help of other modes.

Ir. P. P. Eckersley: Why do so many people turn their speakers up louder than normal speech level? It may be the top frequencies of loudspeaker reproduction are usually ıuated far more than in direct speech, but this suggestion ; not look particularly viable in view of the fact that most ›le turn their tone controls to 'mellow'.

is sad to realize that the B.B.C. takes all this trouble to get ıcoustics of the studio to near perfection while the listener fies all its efforts.

Mr. W. West: For testing loudspeakers, I use the criterion that a source of sound (e.g. a talker), such as could be present in the listening room, should reproduce as though it were in the room. With this criterion any observable effects of the talks studio are unwanted, and thus ideally the studio should be non-reverberant. Recordings of speech that we have made in a non-reverberant room are better, in respect of clarity and signal/noise ratio, than those made in normal rooms acoustically treated as studios. The objection that a speaker does not like talking in a non-reverberant room is unreal; the room has a calm, relaxed atmosphere, and the idea that there is no need to talk loudly is easy to put across. If the talker likes to wear a headphone to hear his voice by sidetone, the loudness of his talking can be controlled, without his knowledge, by adjusting the amount of sidetone he receives.

In Section 5.1 it is stated that binaural hearing reduces the effects of reverberant sound in the listening room. What is called, for brevity, the binaural mechanism is, of course, more than a mechanism, because subtle faculties of the brain are involved. Has the author any experience to indicate whether a so-called stereophonic system, using loudspeakers in the listening room, is of any help in rejecting reverberant sound coming from the studio?

Mr. P. F. Cook: With regard to the intensity of reproduction of a voice from a loudspeaker I suggest that this is largely a psychological matter. In normal conversation our speech content is full of redundancies and the information content is low, but it is very noticeable that, if someone in a group of people conversing in a room says something interesting, the others immediately look at him. This illustrates the fact that we get a lot of information from a speaker present in a room by watching his lip movements and general expressions. When that visible information is removed we feel the need of increased intelligibility. The normal reaction, since the broadcast message usually has a higher information content than ordinary conversation, is to turn up the volume. As soon as the broadcast speaker becomes uninteresting and the people in the room want to discuss things among themselves, the volume tends to be turned down.

I suggest that our normal conversation is conducted at an intensity which is on the borderline of intelligibility and that omissions are made good by the visible factors. When these are absent we choose the increased intelligibility—particularly that associated with the higher frequencies—which result from increased volume.

Major W. V. G. Fuge: It is well known that, if we remove the furniture and carpet from quite a small room, speech has an echo added to it. The less furniture there is, the greater the echo effect. Has any use been made of this phenomenon for adding a pleasing amount of echo to speech, by emitting it from a loudspeaker at one end of such a room and picking it up by a microphone at the other end with the echo added. Can such a desirable added effect be made to predominate over the acoustic effects due to the room in which the speech is made, and to the room in which it is heard?

THE AUTHOR'S REPLY TO THE ABOVE DISCUSSION

r. C. L. S. Gilford (*in reply*): I agree with Mr. Moir that the judgment of a room must be subjective; all the objective : described in the paper was carried out with continual ence to parallel subjective experiments. Most of the :ctive work on listening rooms was carried out on speech, : only speech is normally broadcast from studios having r reverberation times than ordinary living rooms. The stics of the living room would be expected *a priori* to have less importance when listening to programmes emanating from much more reverberant studios, and this is borne out in practice. Measurements of reverberation time at low frequencies have more meaning than he suggests, because repeatable results can be obtained by averaging readings from several microphone positions using a test sound of finite bandwidth and also measuring the decay time of particular modes. I agree that, in practice, flutter presents much less trouble than ringing, but it is not my experience that non-parallel walls reduce ringing.

Mr. Humphreys is right to point out that no studio can be equally correct for all kinds of speech. The requirement which has been borne in mind is that of didactic speech, news bulletins, etc., which comprise the majority of speech programmes. Microphone placing can, to some extent, introduce the perspective required for discussions or the intimacy of a narration. Narrators' studios associated with drama suites are designed to be less reverberant than ordinary talks studios.

With regard to early reflections, there is no doubt that those from a hard table top or a cubicle window too close to the microphone alter speech quality, but I am not convinced of their importance in determining the characteristics of a studio as a whole. There is room for further investigation. An isolated complaint of coloration was actually traced to unsupported plasterboard, and I am glad to know that it is unlikely to happen again.

In reply to Mr. Eckersley, I have found that a subject who is asked to adjust a loudspeaker, so that a colleague's voice reproduced by it appears of the correct loudness, will invariably set the level too high—sometimes by as much as 8–10 dB. I have alv imagined this to result from psychological influences such increased attention, in the case of direct speech, due to physical presence of the speaker. I therefore agree entirely v Mr. Cook's most informative remarks.

Mr. West's statement about the effects of acoustics on speaker conflicts with my experience. The majority of regu and casual broadcasters find a very dead studio unpleasant, a although an inexperienced broadcaster can easily be trained speak quietly, it is better that conditions should be accepta from the start.

Some experiments with a crude stereophonic system carried some years ago were inconclusive. It would be worth while repeat them using better studios and a modern stereopho system.

Major Fuge's idea of using a supplementary room to reverberation is, in fact, in everyday use in broadcasting a recording organizations. The room is usually walled with smo concrete or tiles to give it a long reverberation time, but so absorption may be introduced to adjust the frequency char teristic.

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No. 9

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by C. L. S. GILFORD, M.Sc., F.Inst.P., A.M.I.E.E.
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NOVEMBER 1956

BRITISH BROADCASTING CORPORATION

FOREWORD

THIS is one of a series of Engineering Monographs published by the British Broadcasting Corporation. About six will be produced every year, each dealing with a technical subject within the field of television and sound broadcasting. Each Monograph will describe work that has been done by the Engineering Division of the BBC and will include, where appropriate, a survey of earlier work on the same subject. From time to time the series will include selected reprints of articles by BBC authors that have appeared in technical journals. Papers dealing with general engineering developments in broadcasting may also be included occasionally.

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THE APPLICATION OF PHASE-COHERENT DETECTION AND CORRELATION METHODS TO ROOM ACOUSTICS

SUMMARY

n an attempt to produce an improved method of displaying the acoustic behaviour of a room, an investigation has been nade into some phase-sensitive and correlation methods. In particular, a phase-coherent modification of the pulsed glide ype of display has been developed, in which the microphone output is modulated by the original frequency of excitation efore being applied to the cathode-ray oscillograph. Tests made using this new instrument are described, and the results liscussed. Correlation methods are also discussed, and the results of some tests involving cross-correlation are given. inally, modifications of the phase-coherent pulsed glide, making use of cross-correlation and phase-reversal counting are lescribed.

l. Introduction

n the investigation of the acoustics of a studio, it is very lesirable to obtain a record of its behaviour which shows learly the 'singularities' in its response to sounds of differ-nt frequencies. These singularities may take the form of ong 'rings' at certain frequencies, or of greater or lesser legrees of colouration of the programme, i.e. the undue rominence of certain sharply defined frequencies. Such ffects, which are noticed more commonly in small speech tudios than in larger ones, often appear to be due to imple isolated natural modes of vibration of the air mass n the studio and hence vary greatly in their audibility with he position of the microphone or hearer. This feature enders quantitative study more difficult and instru-nental means of evaluation more attractive.

In order to obtain a permanent record of the transient requency response, the 'Pulsed Glide' type of display was leveloped, as previously described in the *BBC Quarterly*.[1] The basic element of this display is a photographically ecorded time-graph of the decay of the sound pressure in he studio after the cessation of a tone emitted by a loud-peaker. The time axis is along the length of a 35 mm. film nd the pressure axis, normally giving a range of 50 dB, is t right-angles.

The photographic film is moved continuously, and a uccession of traces with slowly increasing tone frequency s recorded, giving a composite display such as that shown ater in Fig. 3. The example given is unusually simple in orm, being the display for a small room at comparatively ow frequencies.

In such displays, colourations due to isolated modes are usually shown as a straightening of the individual traces as at (*a*), owing to the absence of beats with neighbouring modes; the room behaves, in effect, as a simple resonant system with only one mode of vibration. Unfortunately, this feature is not always apparent in the display, particu-larly when there is no accompanying lengthening of the reverberation time, since it may be obscured by irrelevant detail. Moreover at any particular microphone position, the amplitude of a mode which is subjectively important in the room as a whole may be insufficient to give a recognizable feature of the display. It is true that if the dis-play is obtained from several different points the import-ant modes will be revealed but such repetition is too lengthy to be practicable in many cases. Because of these difficulties it was decided to investigate a phase-sensitive method of display which would be less dependent upon the respective amplitudes of the modes.

Changes of pitch always occur during the decay of sound of frequencies on either side of a modal frequency, and pitch changes are also associated with structural resonances. These pitch changes may be recognized by the ear as colourations, if sufficiently prominent, and prob-ably constitute an undesirable subjective feature. In the type of display to be described first, the product of the amplitudes of the decaying sound and of the exciting fre-quency is shown on a logarithmic scale, and this method clearly indicates pitch changes whether these are due to structural resonances or isolated modes.

As with the normal pulsed glide, small studios give more comprehensible patterns than large ones; difficulties arise with large studios mainly because of the extremely rapid variation of response with frequency.

A second possible method of reducing the influence of individual microphone positions and revealing those features which are most generally encountered in the room would be to combine or correlate the outputs of two or more microphones distributed about the room. In Section 4, methods involving cross-correlation are con-sidered and a variation of the phase-coherent display described earlier in the monograph is shown to have some useful features.

2. The Development of the Phase-coherent Pulsed Glide

The principle of the method to be described may be illus-trated by considering the simple case of the sound decay taking place at a single frequency close to the original exciting frequency.

Here the sound decay may be represented by

$$Ae^{-\alpha t}\cos\left\{(\omega+\delta)t+\phi\right\}$$

where A, α, δ, and ϕ are constants, $\omega/2\pi$ is the frequency of the tone, and t is the time. If this is multiplied by the time function of the original tone, $\cos\omega t$, we obtain the expression:

$$\tfrac{1}{2}Ae^{-\alpha t}\left[\cos\left\{(2\omega+\delta)t+\phi\right\}+\cos(\delta t+\phi)\right]$$

The second term in the above expression may be selected by a low-pass filter arranged to remove frequencies greater than, say, 10 c/s. The output then takes the form of an exponentially decaying sine-wave of a frequency equal to the pitch change in the room. When there is no pitch change a true exponential is obtained, whose magnitude and sign depend on the phase of the voltage produced by the microphone.

A conventional ring modulator has been used to achieve the multiplication, and the instrument accordingly responds to odd harmonics of the original frequency in addition to the desired fundamental. It is not possible to remove harmonics by the use of filters in the microphone circuit because of the phase shifts which would be introduced, but no trouble has been experienced in practice.

It is convenient to convert the decays from a linear to a decibel scale, and this logarithmic conversion may be carried out either before or after the multiplication by the original tone. It is very much simpler from an instrumental point of view to pass the returning sound through a logarithmic amplifier before multiplication, but there are theoretical objections, and it has been found in practice that clearer displays more in accordance with prediction are obtained by reversing the order. Another disadvantage of logarithmic conversion before mixing is that this causes the instrument to respond also to odd subharmonics of the original frequency, if these are present in the returning sound. It will be appreciated that, apart from the above considerations, the instrument is virtually insensitive to noise, having an extremely small effective bandwidth.

2.1 *Experimental Chain*

A block schematic diagram showing the normal interconnection of apparatus is given in Fig. 1. An audio frequency signal is obtained from an oscillator and fed simultaneously to the coherent detector and to a 'Tone Pulser', the function of which is to convert the signal into short bursts which are then applied to a loudspeaker in the room under test. The pulses or bursts of tone are long enough to allow steady-state conditions to be reached. The sound picked up by a microphone is amplified and applied to the coherent detector, the output from which is displayed on a cathode-ray oscillograph. The time-base is triggered at the end of the pulse of tone so that only the decaying sound is shown.

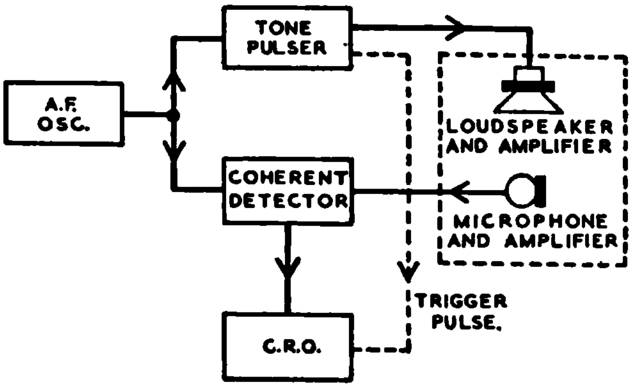

Fig. 1 — Block schematic diagram of coherent detector chain

2.2 *Description of the Instrument*

A simplified circuit diagram of the instrument is given in Fig. 2. It will be seen that two inputs are provided, one for the microphone signal and one for the reference voltage from the oscillator. The reference voltage is limited by the crystal rectifiers X_1 and X_2 and applied to an amplifying stage driving the first ring modulator X_{3-6}.

The microphone output, after amplification by a microphone amplifier (not shown) is applied to the grid of V_2 and thence to the ring modulator X_{3-6} for multiplication by the reference voltage.

The difference frequency is selected from the multiplication products by the low-pass filter L_1, C_1, C_2 and is then superimposed on a 1 kc/s carrier by means of the circuit shown (X_{7-10}) before being applied to a logarithmic amplifying stage V_3. Anti-phase outputs from a multivibrator (not shown) are clipped by the high resistances R_1 and R_2 and the rectifiers X_{7-10}, the resulting square waves being applied to the grid of V_3 through the equal resistances R_3 and R_4. Under conditions of zero output from the low-pass filter the system is symmetrical and no voltage is applied to V_3. However, the presence of a signal unbalances the system, producing a 1 kc/s square wave of amplitude proportional to the output from the filter, and of phase determined by its sign. GEX.66 crystal rectifiers are used as logarithmic elements (X_{11} and X_{12}) and these provide a useful range of at least 50 dB. However, the operating range of the complete instrument is limited by various other effects to about 40 dB.

V_4 drives a second ring modulator X_{13-16}, the switching waveform for which is also derived from the multivibrator. This ring modulator thus functions as a phase-sensitive detector of the output from the logarithmic amplifier. An output of ± 5v. is obtained.

2.3 *Details of the Experimental Method*

It was found quite early in the experimental work that it was essential to interchange the X and Y plates in the oscillograph so that the time base sweep was perpendicular to the direction of motion of the film. If this was not done the patterns obtained were virtually unintelligible. Investigations were also made into the possibilities of normal pulsed glides with the oscillograph plates interchanged in this manner, but here the disadvantages appeared to outweigh the advantages and the idea was not pursued further.

It was also found that an optimum gain exists for the microphone amplifier driving the coherent detector. With excessive gain extraneous detail is liable to be recorded, while useful information may be lost if the gain is too low.

In general, it is desirable to include the maximum number of individual decays in the display, as this enables small formations to be identified more easily. A frequency range of 50 c/s to 500 c/s has been used in the investigations, as above 500 c/s the change in frequency of the oscillator during the period of a decay becomes significant. A glide rate of 8 minutes per octave was found to be suitable for most purposes. Oscillator stability becomes a serious problem for slower glides than this and the time required becomes excessive.

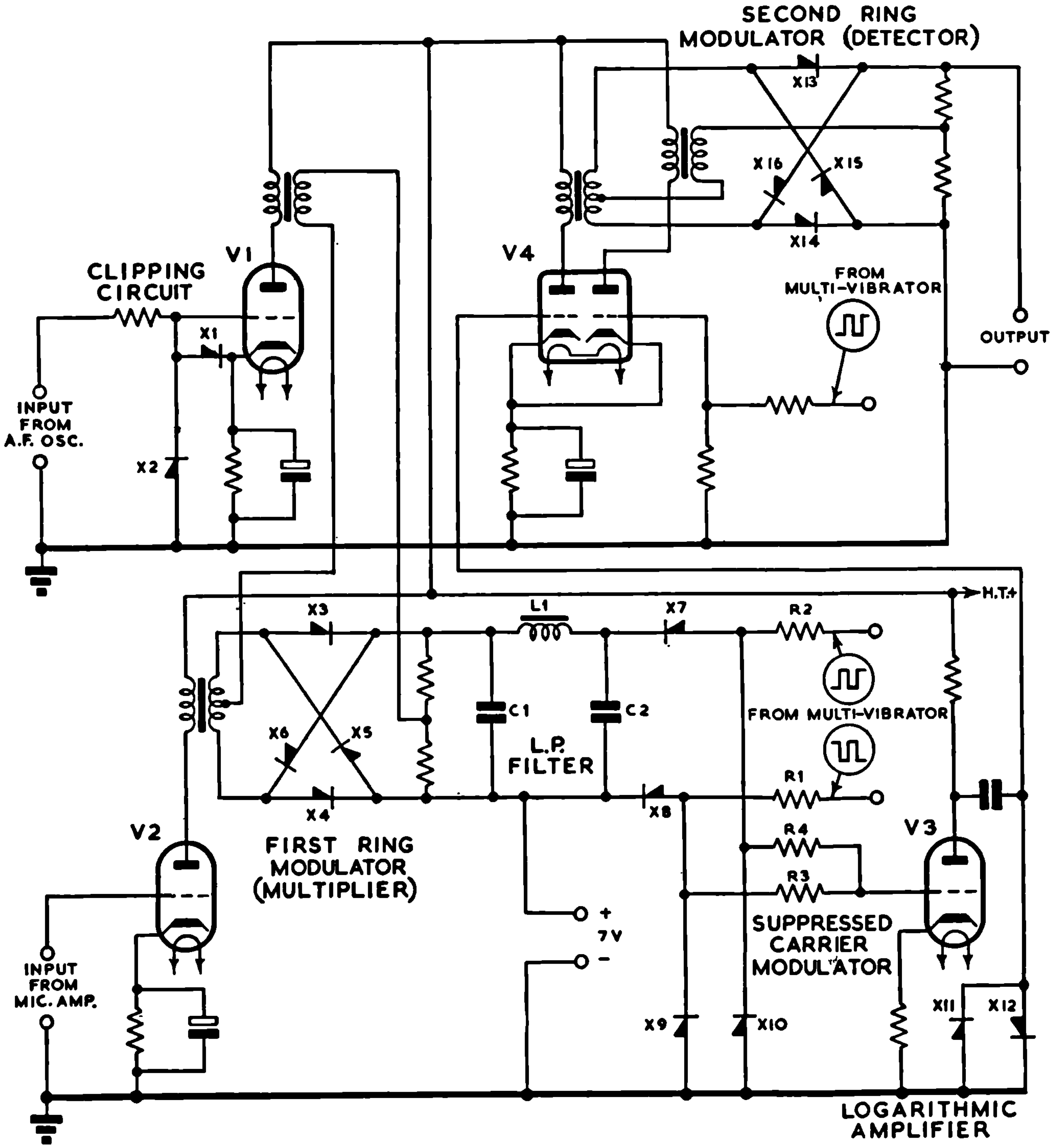

Fig. 2 — Simplified circuit diagram of phase-coherent detector

3. Experimental Work and Results

The experimental work undertaken was divided into three main groups:

(*a*) Coherent glides of rooms.

(*b*) Coherent glides of electrical networks. This was undertaken to investigate the extent to which room behaviour can be simulated by electrical networks.

(*c*) Investigations in rooms using phase-reversal counting.

3.1 *Investigations in Rooms*

Most of the experimental work was done in a small experimental talks studio in the Research Department. It has dimensions 15 ft 7½ in. × 12 ft 8 in. × 9 ft 6 in. (4·77 × 3·86 × 2·90m) and a volume of 1 880 cu. ft (54m³). Table I gives a list of the frequencies of the axial modes for this studio.

TABLE I

Table of Axial Modes up to 500 c/s for Experimental Talks Studio

c/s	c/s	c/s	c/s
36·2	144·6	267·5	397·7
44·6	178·3	289·2	401·2
59·5	178·3	297·3	416·3
72·2	180·8	312·0	433·9
89·1	216·9	325·3	445·8
108·4	223·0	356·7	470·0
118·9	237·8	356·8	475·9
133·7	253·1	361·6	490·5

A series of tests was undertaken to compare the coherent glide with the normal pulsed glide as a means of indicating the frequencies of the principal subjective colourations. Listening tests with speech showed the latter to lie at 90, 140, 175, and 220 c/s, and it will be seen from the table that these correspond to axial room modes or groups of modes.

Early in the tests it was established that no satisfactory correlation was obtained between pulsed glide displays and colourations unless the loudspeaker used for producing tone for the displays was comparable in size with the source used for the subjective tests, i.e. the human head. An '8-inch' loudspeaker unit was therefore used in a cabinet of only 25 cm. square frontal area. Six coherent displays, from different microphone positions, and five normal pulsed glide displays were examined by three observers and the frequencies of indicated singularities were listed. Corresponding formations of the types associated with strong isolated modes are visible in Figs. 3 and 4. The straight line formation of the normal pulsed glide appears at (*a*) in Fig. 3, while at (*b*) in Fig. 4 is the corresponding pattern obtained by the coherent detection method. The general slope of the decays in the latter is the cause of the noticeable asymmetry of the pattern, and this would be reversed if the microphone leads were interchanged.

The results of these tests may be summarized as follows:

(i) *Normal Pulsed Glide*
This showed severe colourations at 137, 160, and 225 c/s with a minor group at 150 c/s.

(ii) *Coherent Display*
The most severe formations shown were in the range 215–225 c/s, other sharply localized groups being at 88–98 c/s, 140–150 c/s and 320 c/s. There was also a diffuse group ranging from 180 c/s to 200 c/s.

Thus the normal display showed two of the subjectively important colourations and two spurious frequencies; the coherent display showed three of the known colourations with two spurious groups. It may be remarked that the coherent display showed the very important colourations at nearly all microphone positions, whereas the conventional display was less uniform in this respect.

The coherent display, therefore, appears to be slightly better as a means of diagnosis, although it shares the disadvantages that several microphone positions are necessary and that spurious indications are normally present as well as correct ones.

An examination of a typical coherent glide shows that simple, clear patterns are only rarely obtained. Most of the formations present are of a random nature, probably due to reflections taking place before standing wave systems can be built up. These effects tend to obscure any regular patterns in the display.

It will be noticed from Fig. 4 that compression or bunching of the commencement of the traces, i.e. at the bottom, occurs at intervals along the display. This is due to the progressive phase advancement with frequency which takes place with a fixed microphone and loudspeaker spacing. (See Section 3.3 below.) Patterns char-

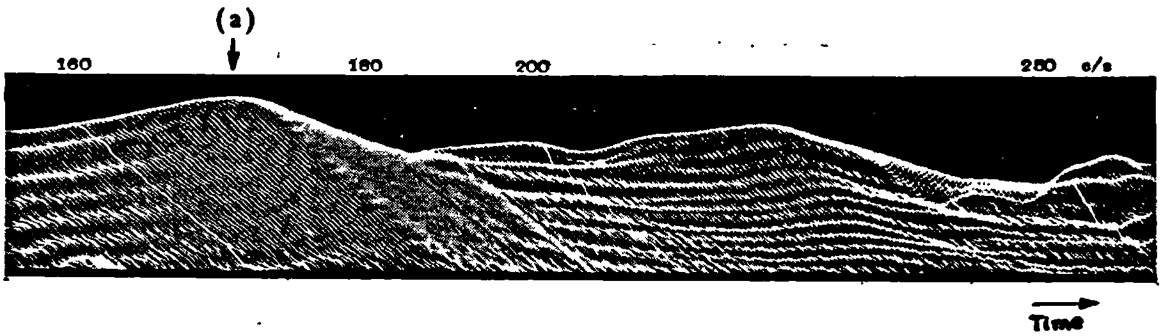

Fig. 3 — Part of a normal pulsed glide of a small room

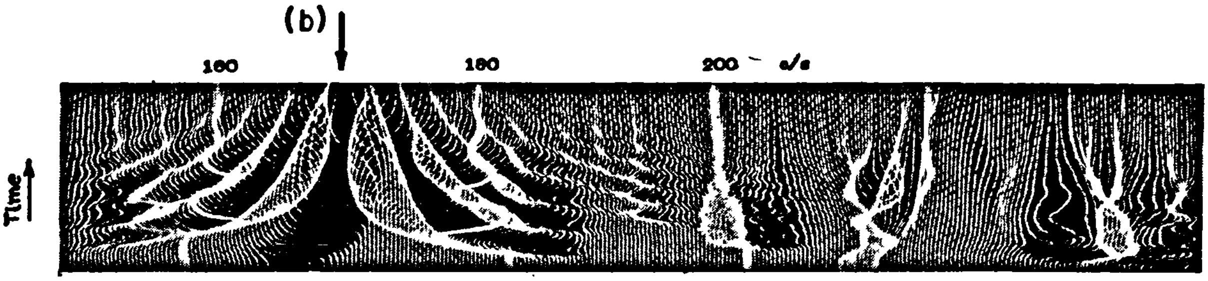

Fig. 4 — Part of the phase-coherent pulsed glide corresponding to Fig. 3

acterized by a beat frequency which is constant and independent of the oscillator frequency over a region of several cycles may also be observed occasionally in the display. These beats probably take place in the room itself between adjacent modes, and are not a result of the modulation process in the coherent detector.

Fig. 5 is part of a coherent glide taken in a large orchestral studio (Maida Vale Studio No. 1). It will be seen that successive decays differ too much for any patterns to be visible. This is because the response of the studio changes very rapidly with frequency. In addition the longer reverberation time of the studio makes it necessary to allow a greater interval between successive pulses and therefore for a given rate of frequency glide differences between the frequencies of consecutive pulses are greater than for small studios. To avoid this effect, an impracticably slow rate of frequency glide would be necessary.

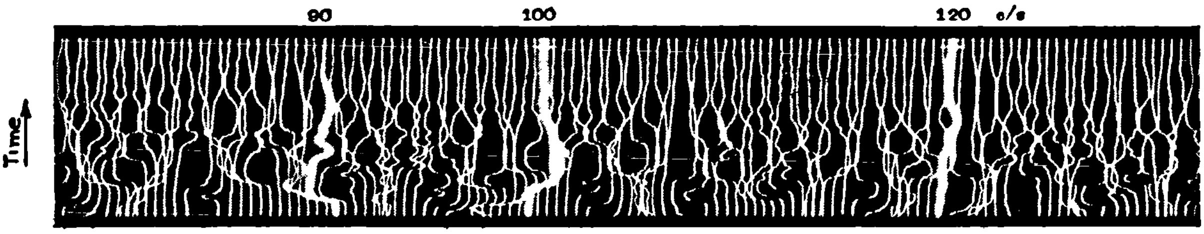

Fig. 5 — Part of a phase-coherent pulsed glide taken in a large studio (Maida Vale 1)

3.2 *Some Theoretical Considerations in Connection with Small Rooms*

Compared with the conventional pulsed glide, the coherent type of display possesses a slightly superior resolving power. This is illustrated by one particular case in which a region only a few cycles wide was shown by the coherent display to contain two very closely spaced modes. The region appeared on a conventional pulsed glide simply as a series of smoothly rounded decays.

The improved resolution is largely due to the fact that there is a visible change of phase as a room mode is passed through. If we consider two modes 2 c/s apart in frequency, it will be appreciated that they will not come into anti-phase until 0·25 second has elapsed. Therefore in a display sensitive only to amplitude there will be no clear indication of beats unless the time of sweep across the oscillograph exceeds this time, that is unless the reverberation time of the room exceeds 0·30 second. Even when this condition is satisfied, it is necessary to ensure that the frequency change between pulses is not greater than 1 c/s if it is desired to preserve this detailed information. This remark applies, of course, equally to both types of display. It should be noted that since data concerning normal modes are, in general, most useful in connection with colourations in small talks studios, and as these rarely have reverberation times in excess of 0·4 second, the coherent type of display possesses an intrinsic advantage in this application.

3.3 *Coherent Glides of Electrical Networks*

In order to obtain a better understanding of coherent glide displays, from rooms, it was decided to investigate the displays obtained from certain simple networks, such as tuned circuits. The essentially resonant nature of room modes suggested that patterns similar to those obtained from strong isolated modes might be obtained from tuned circuits. This was in fact shown to be the case.

Coherent pulsed glides were taken of the following networks:

(*a*) Single tuned circuit. $Q=50$, $f_r=250$ c/s approx.

(*b*) Two tuned circuits, uncoupled and with various spacings between their frequencies of resonance. $Q=50, f_r=250$ c/s approx.

(*c*) All-pass network.

The response of the coherent detector to a single tuned circuit was also calculated and the results compared with those obtained in practice. To facilitate the comparison of individual decays, the spacing between the decays was increased in two of the glides. The glides of the single tuned circuit are shown in Figs. 6, 7, and 8, and some of the computed curves are given in Fig. 9.

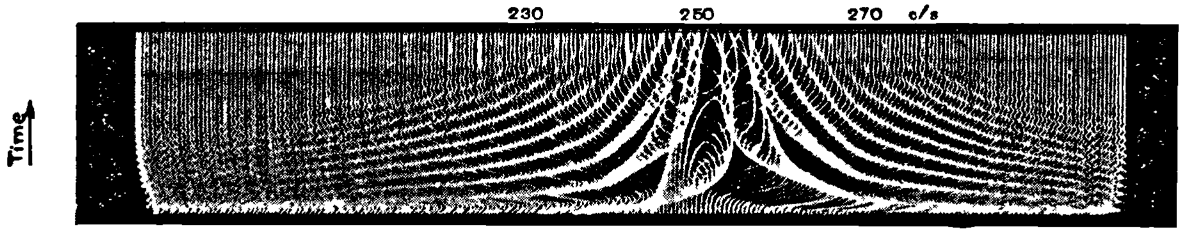

Fig. 6 — The pattern produced by a phase-coherent pulsed glide of a tuned circuit

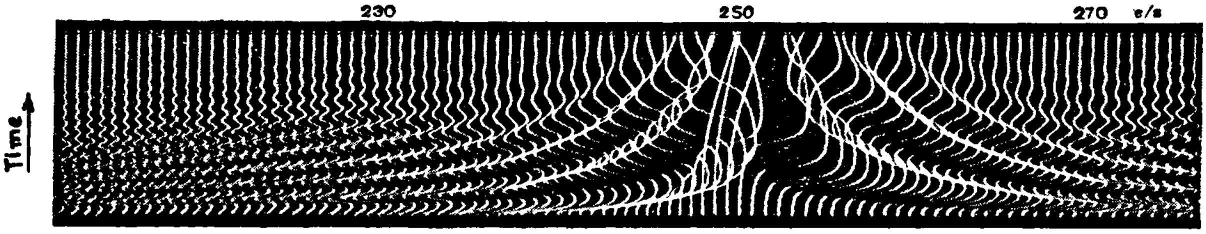

Fig. 7 — The pattern produced by the tuned circuit of Fig. 6, with the frequency scale expanded

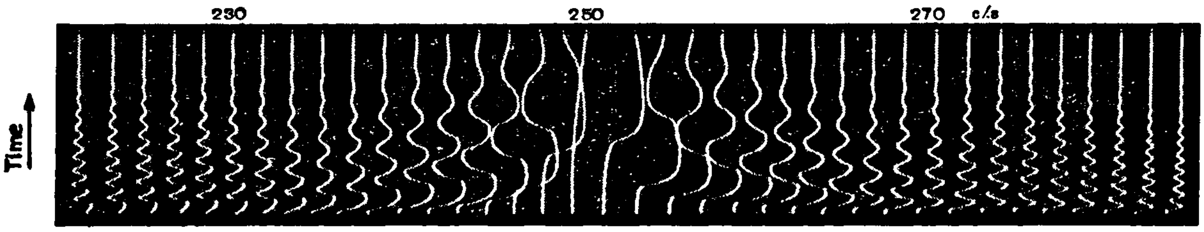

Fig. 8 — As Fig. 7, but with the pulsing rate reduced to enable individual traces to be seen

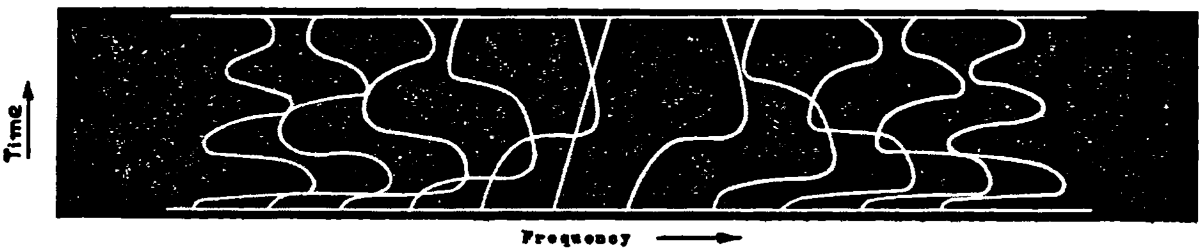

Fig. 9 — Computed curves for the single tuned circuit, for comparison with Fig. 8

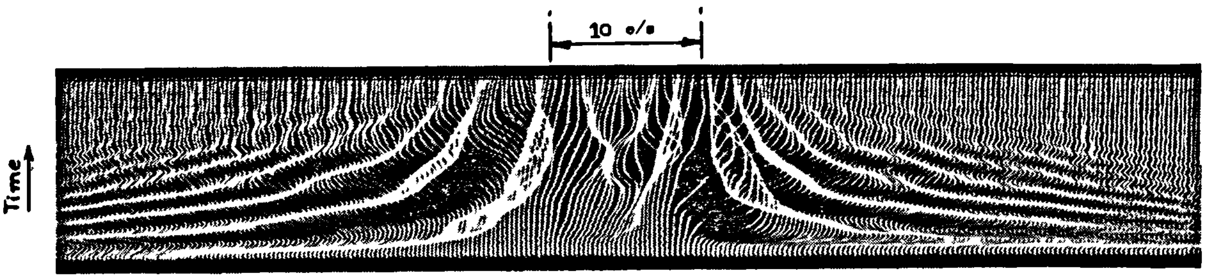

Fig. 10 — *Part of a phase-coherent pulsed glide of two tuned circuits, uncoupled, and with a resonance frequency separation of* 10 c/s

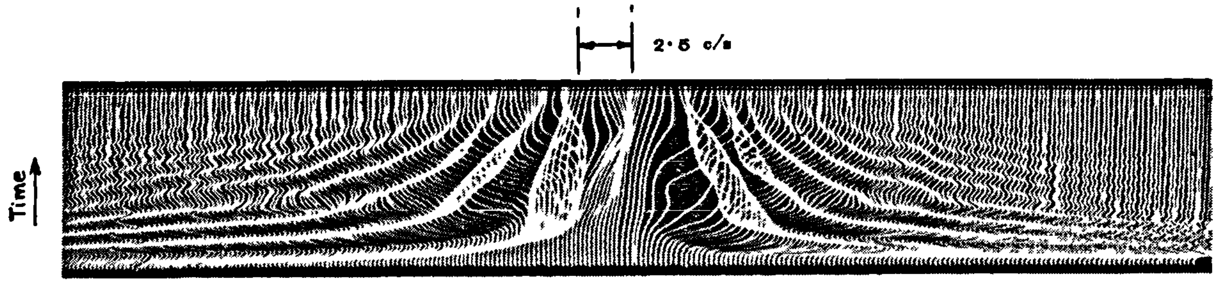

Fig. 11 — *Part of a phase-coherent pulsed glide of two tuned circuits, uncoupled, and with a resonance frequency separation of* 2·5 c/s

Figs. 10 and 11 show the response of the coherent detector to the two tuned circuits (see (*b*) above), with resonance separations of 10 c/s and 2·5 c/s. The glide rate of the oscillator was specially reduced in this case, to lessen the complexity of the patterns produced. It will be seen by comparison with Fig. 6 that resonance separations of 2–3 c/s are resolvable.

Part of a 'glide' of the all-pass network shown in Fig. 12 is given in Fig. 13. It illustrates the bunching and expansion of the decays which result from a phase characteristic $\phi(\omega)$ of the form

$$\phi(\omega)=2N\tan^{-1}k\frac{\omega}{\omega_r}\left\{1-\frac{\omega^2}{\omega_r^2}\right\}^{-1}$$

where N is the number of sections of the network
k is a circuit parameter
ω_r is the frequency at which $\phi=N\pi$

A similar effect of dispersion is observable in all coherent glides of rooms; it appears as a background to the more striking patterns due to eigentones and strong early reflections.

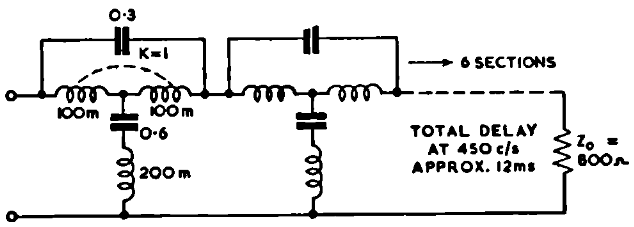

Fig. 12 — *Circuit diagram of all-pass network*

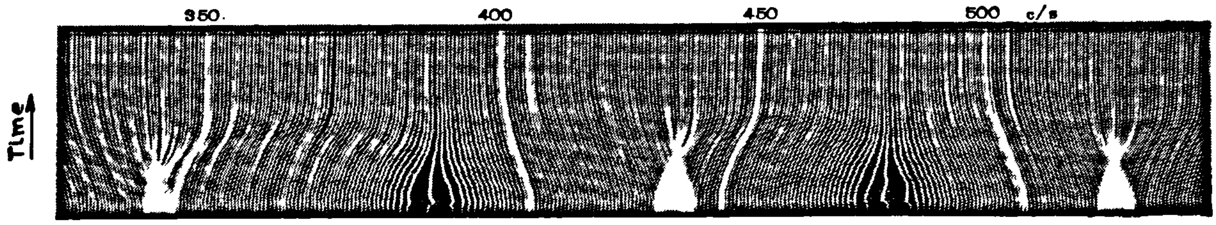

Fig. 13 — *Part of the phase-coherent display of the network illustrated in Fig.* 12, *showing effects of periodic phase reversal*

3.4 *Phase-reversal Counting*

As an alternative means of detecting colouration frequencies, slight modifications have been made to the output stage of the coherent detector so that it can be used to operate a counter which indicates the number of phase reversals during the decay of the sound.

When the frequency of the exciting tone is equal to the frequency of a normal mode of the room, there is no pitch change and no count is obtained. As the frequency of the exciting tone is moved away from that of the normal mode, the count for each pulse gradually increases until the normal mode is no longer excited and the count suddenly ceases.

The microphone is placed in one corner of the room, with the loudspeaker in another corner, a corner being the best position since all room modes are there equally excited. Two values of the count are taken for each microphone position, one with the microphone connections reversed in order to eliminate any asymmetry in the apparatus. The colouration frequencies, as indicated by the counter, were found to be almost independent of the corners chosen for the loudspeaker and microphone though the magnitude of the count did vary.

4. Methods Involving Auto- and Cross-correlation

A study of methods involving correlation was made in an attempt to measure properties that are characteristic of the studio as a whole, rather than of particular microphone positions. As mentioned in the previous section, this is one of the fundamental difficulties in acoustical measurement, and correlation methods would appear to be very advantageous in this respect.

Gershman[2] has used correlation methods in an attempt to measure a quantity corresponding to the liveness of a room.

If $v(t)$ is the instantaneous sound pressure at time t at a point in the room, and τ any arbitrary time interval, the auto-correlation function of $v(t)$ may be written in its simplest form as

$$\psi(\tau)=\lim_{T\to\infty}\frac{1}{2T}\int_{-T}^{+T} v(t)v(t+\tau)dt$$

Parseval's theorem[3] states that if $f_1(\omega)$, $f_2(\omega)$ are the Fourier transforms of $\phi_1(t)$, $\phi_2(t)$, and if * indicates the conjugate of a function of a complex variable,

$$\frac{1}{2\pi}\int_{-\infty}^{+\infty}\phi_1(t)\phi_2(t)dt=\int_{-\infty}^{+\infty}f_1^*(\omega)f_2(\omega)d\omega=\int_{-\infty}^{+\infty}f_1(\omega)f_2^*(\omega)d\omega$$

Writing $f(\omega,t)$ as the Fourier transform of $v(t)$ we see that $f(\omega, t)e^{i\omega\tau}$ is the transform of $v(t+\tau)$ and hence the auto-correlation function may be written

$$\int_{-\infty}^{+\infty}\lim_{T\to\infty}\frac{\pi}{T}\left|f(\omega, T)\right|^2 e^{i\omega\tau}d\omega,$$

since the function $f(\omega,T)$ is real and equal therefore to its conjugate.

Since $|f(\omega,T)|^2$ is an even function of ω,

$$\psi(\tau)=\int_0^{\infty}\lim_{T\to\infty}\frac{2\pi}{T}\left|f(\omega, T)\right|^2\cos\omega\tau d\omega$$

We may write

$$F(\omega)=\lim_{T\to\infty}\frac{2\pi}{T}\left|F(\omega, T)\right|^2$$

where $F(\omega)$ is the spectrum power function, i.e. the energy density at frequency $\omega/2\pi$ and hence $\psi(\tau)$ is given by

$$\int_0^{\infty}F(\omega)\cos\omega\tau d\omega$$

Gershman expresses this in a normalized form

$R(\tau)=\frac{1}{V^2}\int_0^{\infty}F(\omega)\cos\omega\tau d\omega$ where V is the r.m.s. amplitude,

and shows that in general, the function falls substantially to zero after an interval τ_0 which he defines as the 'Coherence interval'.

He prefers to express this interval as the distance $c\tau_0$ travelled by the sound from the source. For pure tone $c\tau_0$ is infinite, but a band of noise of finite width gives a finite correlation interval which decreases as the bandwidth increases. Thus, an octave band of noise from 800 c/s to 1 600 c/s has a coherence interval of about 70 cm, and the band from 3 200 c/s to 5 400 c/s only about 18 cm. Outside these distances any observed correlation between the signal from a microphone and the exciting signal from the loudspeaker must be due to standing wave effects, and any cross-correlation observed between two microphones at a similar distance apart must be due to standing wave effects or symmetry with respect to the source.

If $v(t)$, $u(t)$ are the two microphone signals in the latter case, the cross-correlation between them may be expressed most generally as

$$\lim_{T\to\infty}\frac{1}{2T}\int_{-T}^{+T} v(t)u(t+\tau)dt$$

The effect of varying τ is not significant for the present purpose and therefore Gershman makes $\tau=0$, giving

$$\lim_{T\to\infty}\frac{1}{2T}\int_{-T}^{+T} v(t)u(t)dt$$

which is simply the time-averaged product of the two instantaneous signals. He shows that in the case of two microphones placed symmetrically about the axis of a loudspeaker, this function is a useful measure of the liveness of a room.

It will be seen from this that Gershman used the departure from complete correlation in a symmetrical case to indicate a phenomenon, viz. liveness, due to reflections from the walls of the enclosure. If, instead, the symmetry is eliminated and the distances between the transducers are all made greater than the coherence interval, there will be zero correlation unless there are appreciable standing wave effects at both positions, since standing wave systems have only two (opposite) values of phase throughout the room.

4.1 *Cross-correlation Methods*

This approach has been followed in some investigations in two experimental talks studios. The circuit used for performing the cross-correlation consisted of a ring modulator fed with the two microphone signals, each of which was adjusted independently to a known fixed value. This adjustment having been made, the output from the modulator gave the correlation coefficient. A diagram of the circuit is given in Fig. 14. Uncorrelated inputs from two noise generators give a coefficient of less than 0·1 by this method, and this figure was taken as the threshold of accuracy of the method. Microphones were placed at two points in the room, and a loudspeaker was fed with a band of noise wide enough to put both microphones outside the coherence interval with respect to each other and the loudspeaker. A difficulty arises here, however, since the coherence interval for a talks studio of normal size may be greater than the room dimensions except for bands of noise wide enough to excite many modes. Each standing wave system will give either +1 or −1 correlation between two microphones, and hence the coefficient for the band may have any value between −1 and +1, depending upon the number and sense of the separate correlations within the band. In the actual experiments two pressure microphones were placed in each of the twelve pairs of corners having diagonal relationships to each other (i.e. two or three co-ordinates different), the loudspeaker being in another corner. For each pair the correlation coefficients were read for octave bands of noise with central frequencies ranging from 120 c/s to 1 700 c/s. The mean of the moduli of the twelve figures for each band would be a measure of the importance of the standing wave effects in the band.

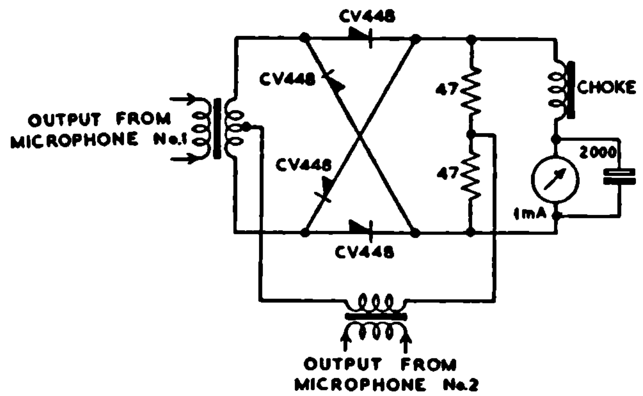

Fig. 14 — Circuit diagram of arrangement for cross-correlation experiment

The results were as follows:

TABLE II

Mean Cross-correlation Coefficients between Pairs of Microphones in a Studio

Mid-frequency	120	240	480	950	1 700 c/s
Studio No. 1	0·26	0·21	0·12	0·11	0·05
Studio No. 2	0·27	0·25	0·10	0·10	0·08

The differences between the two studios are thus quite insignificant, and it is concluded that the figures obtained are a function more of the method than of the studio. No useful information as to the characteristics of the studio could therefore be obtained in this way.

4.2 *Coherent Glide between two Points in a Studio*

Another possible method of obtaining a synthesis of two microphone positions is to use a modified form of the coherent detection display described in Section 2. The normal coherent detection equipment is used but the tone input to the modulator is replaced by a signal from a second microphone, the output of which has been amplified and limited. This method gives a simpler display than the single-microphone method previously described, because the beats of continuously varying frequency, which form a prominent feature of the latter, are absent.

Fig. 15 shows part of one such display, using microphones in two corners of a small room. It will be seen that the display consists mainly of comparatively unmodulated decays interspersed with regions where the decays show beats of constant frequency over an appreciable range of exciting frequency. Each of the two microphones, being in a corner of the room, will be excited at all modal frequencies, and at any particular exciting frequency the modes most strongly excited will be those adjacent in frequency above and below. Hence the display will tend to consist of straight lines at or around the modal frequencies,

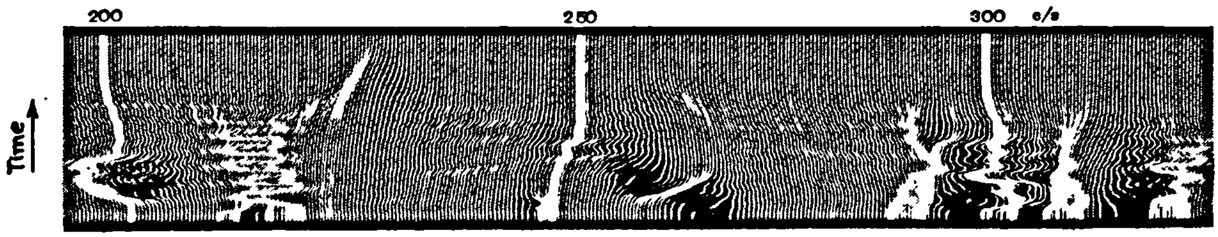

Fig. 15 — Part of a cross-correlated coherent pulsed glide taken in a small room

while at frequencies approximately midway between them, there will be strong beats. The substitution of a signal from the room for the tone used in the ordinary coherent display causes the disappearance of the beats of continuously varying frequency which are a feature of the frequency region surrounding isolated eigentones in the ordinary display. This method therefore has possibilities as a means of exploring the distribution of particular room modes but the possibilities have not yet been exploited.

4.3 *Cross-correlation Methods in the Measurement of Sound Insulation*

Two further applications of auto-correlation and cross-correlation may be of interest here, although they are not directly related to the work described above.

It is often desirable to estimate the sound insulation between two adjacent rooms of a building which is in course of construction or alteration. It often occurs in these cases that owing to the incomplete state of the building, there are indirect paths by which sound can be transmitted between the two rooms without traversing the partition wall which in the final state will be the easiest path. Similar problems occur also in finished buildings. The usual method of sound transmission measurement is to place in one room a loudspeaker radiating a suitable test sound and to measure the intensities in the two rooms in turn. The transmission loss figure thus obtained represents the total effect of all possible paths.

The contributions of individual paths may be measured if their times of transmission can be taken into account. Raes[4] has proposed the use of short pulses of tone as the test sound, the transmitted signal being displayed on a cathode-ray oscilloscope with a rapidly moving time-base. The sound transmitted by the several paths will then be indicated by a series of deflections, the amplitudes of which represent the relative contributions.

This method is rather better suited to high frequencies than low since the minimum length of pulse of which the frequency can be accurately enough determined is about 3 cycles, and at, say, 100 c/s the corresponding time discrimination would therefore be limited to about 30 milliseconds, i.e. 33 ft of path. In broadcasting studios, the frequency range in which adequate sound insulation is difficult to obtain is mainly below 100 c/s.

A method described by Goff[5] has therefore been put into use, in which the cross-correlation coefficient is derived between a microphone in the loudspeaker room and one in the receiving room, an adjustable delay being inserted into the former to compensate for the transmission time to the latter.

The cross-correlation coefficient is calculated automatically by electronic multiplying circuits and its current average value over a short period is indicated on a direct reading meter.

If we thus compare the sound at two microphone positions, one close to the loudspeaker and the other a few feet away, the instrument will show a very high cross-correlation coefficient, provided that a delay equivalent to the path difference between them is introduced into the signal from the nearer microphone. If the two microphones are on opposite sides of a wall, the correlation coefficient is again high, though the signal from the remote microphone is greatly reduced. The increase of amplifier gain in this chain required to restore the product of the two signals to the 'no-wall' value is a fairly accurate indication of the attenuation of the sound along the direct path through the wall. If there is an alternative path, the delay may be increased to correspond to this transit time, and the attenuation again measured.

Another application suggested by the same author is to the identification of a source of disturbing noise from a number of possibilities. For instance, intermittent low-frequency noise was observed in the site for a new studio in a large building; this could have been caused by one of two lifts, by ventilation plant, by heavy traffic near the building or by one of several less likely causes. The noise was finally traced to traffic but the elimination of the other possibilities was a lengthy process, which could be done only at night when there was little activity in the building. For such cases it would be possible to take a microphone into the vicinity of each of the suspected noise sources and measure the correlation coefficient obtainable after adjusting the delay to obtain a maximum figure.

The source giving the highest coefficient is the most probable cause of the interfering noise.

5. Conclusions

The work which is described in the first part of this monograph is an attempt to improve upon the Pulsed Glide display as a method of diagnosis of colourations in small studios. It has been shown that the addition of information about the relative phases of the input and reverberant sounds enables the important room modes to be identified with greater certainty, and the addition of a phase-reversal counter has increased the effectiveness of the method. A further advantage of the introduction of phase information is that it is possible to resolve modes separated by smaller frequency intervals than hitherto.

Auto- and cross-correlation methods of diagnosis have also been examined, but show less promise. Applications to sound insulation measurement are, however, being tested with some success.

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Helmholtz resonators in the acoustic treatment of broadcasting studios

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A theory of the action of Helmholtz resonators as sound absorbers is presented, covering both the isolated resonator and regular arrays. Experiments in reverberation rooms and acoustically treated studios are described and general recommendations for design are given. Regular arrays are preferable to single resonators, openings being made more resistive by covering with a fabric. It is concluded that great variations in design to suit architectural requirements may be made without loss of effectiveness, and the widths of the frequency band over which absorption takes place may be varied between wide limits.

1. INTRODUCTION

Helmholtz resonator consists of a cavity which has a aratively narrow neck. An alternating air pressure ed to the opening of the neck will cause the air in the to oscillate, the natural frequency of the oscillation determined by the mass of the air in the neck and the ess presented by the air enclosed behind it. Any such ation is accompanied by viscous losses, particularly in eck where the particle velocity is highest, and the Helmholtz resonator therefore acts as a sound absorber, the absorption being greatest at the resonance frequency, at which the particle velocities are highest.

The use of Helmholtz resonators as sound absorbers in architectural acoustics offers great advantages since it is possible to design quite small resonators to absorb efficiently at very low frequencies. They therefore provide an alternative in the extreme low frequency region to bulky and costly porous absorbers on the one hand and, on the other, to wood panelling, the absorption of which is not amenable to accurate prediction.

Another effect, however, may be present; the rate of energy loss may in some circumstances be so low in relation to the total stored energy in the system that the resonator will have the effect of prolonging the reverberation time of a heavily damped room. It has been suggested[1] that pots embedded in the walls of ancient Scandinavian churches may have served to enhance the traditional ecclesiastical acoustics, not to improve intelligibility of speech by reducing the reverberation time, as had been assumed previously. More recently resonators of this kind have been used in Denmark for lecture halls and broadcasting studios. Although the derivation of the resonance frequency is well known, there is no accepted theory to guide the use of Helmholtz resonators in room acoustics, recent papers having presented analyses and numerical results which cannot be reconciled.

This paper proposes a theory of the behaviour of such resonators as sound absorbers and describes experiments made to verify it. The experiments were limited to frequencies below 200 c/s, as sound of higher frequencies is more conveniently absorbed by porous absorbers. Applications to higher frequencies have been discussed by other authors.

2. THEORY

2.1. *Determination of resonance frequencies.*

The complete series of modes of vibration is best studied by considering the system as two cylindrical pipes, the neck forming one and the cavity the other (cf. Richardson[2]).

Let ρ, η = density and viscosity of air respectively
r, S, l_0 = radius, cross-section and length of neck respectively
l = effective length of neck. For resonators of small neck diameter compared with cross-section of cavity, $l = l_0 + 1{\cdot}7r$
$m = Sl\rho$ = effective mass of air in neck
S', l', U = cross-section, length and volume of cavity
$\omega/2\pi$ = frequency of incident sound
c = velocity of sound in air

Let Z_0, Z_1, and Z_2 be the impedances at the positions shown in Fig. 1.

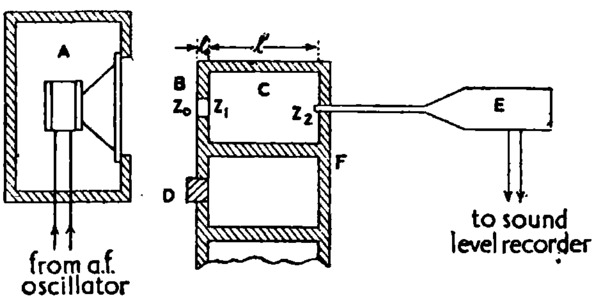


Fig. 1. Arrangement used to obtain response curves
A, loudspeaker; *B*, neck; *C*, cavity; *D*, corked hole; *E*, probe microphone; *F*, block of resonators.

The characteristic acoustic impedance of a tube of cross-section S is $\rho c/S$. We may therefore write the value of Z_1 in two forms as follows:

$$Z_1 = \frac{\frac{i\rho c}{S}\left(Z_0 - \frac{i\rho c}{S}\tan\frac{\omega l}{c}\right)}{Z_0 \tan\frac{\omega l}{c} - \frac{i\rho c}{S}} = \frac{\frac{i\rho c}{S'}\left(Z_2 - \frac{i\rho c}{S'}\tan\frac{\omega l'}{c}\right)}{Z_2 \tan\frac{\omega l'}{c} - \frac{i\rho c}{S'}} \quad (1)$$

At resonance Z_0 must be zero (neglecting losses) and si Z_2 is infinite the equations reduce to

$$\tan\frac{\omega l}{c}\tan\frac{\omega l'}{c} = \frac{S}{S'}$$

The roots of this equation are the values of ω for resonan In practice, three special cases only are of interest.

Case 1. $\omega l, \omega l' \ll c$.

Here $\omega^2 ll'/c^2 = S/S'$, which has only one positive ro giving

$$f = \frac{\omega}{2\pi} = \frac{c}{2\pi}\sqrt{\frac{S}{lU}}$$

Case 2. $S = S'$ (stopped pipe).

$$\tan\frac{\omega l}{c}\tan\frac{\omega l'}{c} = 1 \qquad \text{which gives}$$

$$\omega(l + l')/c = (2n + 1)\,\pi/2$$

Whence
$$f = \frac{2n + 1}{4(l + l')}c$$

Case 3. $\omega l \ll c$ (long resonator with short neck).

Here
$$\omega \tan \omega l'/c = cS/lS' \quad ($$

This equation has an infinite series of roots, and graphi solution shows that the first is close to that of Case 1, and t rest approximately equal to those of a pipe of length l' clo at both ends, i.e. $\omega = \pi nc/l'$. The first root is very mu lower than any of the subsequent ones; for example, in o of the experimental resonators described below the roots a

$$f = 75, 950, 1\,900 \ldots \text{c/s}$$

This wide separation has an advantage for the prese purpose, since by suitable design the absorption at the high modes may be made negligible.

2.2. *The Helmholtz resonator as a simple resonant system.*

Resonators satisfying Case 3 may be regarded, over a wi frequency band on either side of the lowest mode, as a simp system with one degree of freedom. The relevant parametei are then

$$\text{acoustic inertance} = m/S^2 = l\rho/S \quad ($$

and
$$\text{acoustic compliance} = U/\rho c^2 \quad ($$

The acoustic resistance has two components, the radiatio resistance R_1 and the internal resistance R_2, due to visco flow in the neck and other sources of loss. The intern resistance is determined solely by the properties of the nec but the radiation resistance varies according to the nature the sound-field and the proximity of other resonators. For single resonator in an infinite wall,[3]

$$R_1 = 2\pi\rho c/\lambda^2 \quad ($$

and for one of an infinite equally spaced two-dimensiona array

$$R_1 = \rho c/\sigma \quad (9$$

where σ is the area of wall per resonator.

For an open cylindrical neck[3]

$$R_2 = \frac{l}{rS}\sqrt{(2\rho\eta\omega)} \quad (10$$

Finally, the "magnification" of the system, Q, is given by

$$Q = \frac{l\rho\omega}{S(R_1 + R_2)} \quad (11$$

Absorption of sound.

the inside of the resonator is rigid and non-porous, the rption of energy during excitation will occur entirely in neck. The viscous resistance of the neck, given by tion (10), may be increased by altering the neck dimen- ; or by covering or filling the neck with a fibrous or us material.

: resonance frequency the reactance of the resonator is and hence the volume flow in the neck is $p/(R_1 + R_2)$, e p is the r.m.s. pressure. The rate of absorption of gy is, therefore, $R_2[p/(R_1 + R_2)]^2$ which attains its maxi- ı value when $R_1 = R_2$. The problem of achieving the :est possible sound absorption reduces to one of matching nternal resistance of the neck to its radiation resistance. $_2 = \mu R_1$, the efficiency of the neck as an absorber com- d with the value of $\mu = 1$ is $4\mu/(1 + \mu)^2$.

vo important cases must be examined: that of a single ıator in an isolated position, equation (8), and in a plane /, consisting of a large number of resonators equally ed over a plane surface, equation (9). The radiation tances differ greatly in these two cases.

3.1. *Single resonator.* In this case, we have $R_2 -$ $/\lambda^2$ and hence the rate of absorption of the matched lition, where $R_1 - R_2$, is

$$\frac{p^2}{4R_1} = \frac{p^2\lambda^2}{8\pi\rho c}$$

e equate this to the energy flow in the tube of area A' inating in the wall surface, we may say that the absorption uivalent to 100% over an area A' surrounding the neck. energy flow is given by $(p^2A')/(\rho c)$ and equating this to ıbsorption gives $A' = \lambda^2/8\pi$.

or any other value of R_2, where $R_2 = \mu R_1$,

$$A' = \frac{\lambda^2}{8\pi}\frac{4\mu}{(1 + \mu)^2} \tag{12}$$

his quantity A' is the number of absorption units to be :d to the denominator of Sabine's expression for reverbera- time. Brüel points out[1] that the presence of a wall ounding the resonator neck will almost double the sure so that A' has a theoretical upper limit of

$$\frac{\lambda^2}{2\pi}\frac{4\mu}{(1 + \mu)^2} \tag{13}$$

3.2. *Resonator in an array.* Here the radiation resistance :$/\sigma$ and hence the energy absorbed per second is given by

$$\frac{p^2\sigma}{4\rho c}\cdot\frac{4\mu}{(1 + \mu)^2}$$

refore since

$$\frac{p^2A'}{\rho c} = \frac{p^2\sigma}{4\rho c}\cdot\frac{4\mu}{(1 + \mu)^2}$$

$$A' = \frac{\sigma}{4}\cdot\frac{4\mu}{(1 + \mu)^2} \tag{14}$$

he array is embedded in a wall, so that the pressure is bled, this becomes

$$A' = \frac{4\sigma\mu}{(1 + \mu)^2} \tag{15}$$

wing that at resonance a maximum absorption coefficient ınity may be obtained over the whole area of the array $\mu = 1$.

2.4. *The effect of stored energy.*

In a very important paper Rschevkin[4] showed that the simple theory of absorption outlined above was inadequate for application to room acoustics, ignoring as it does the effect of energy storage in the resonator, which may actually increase the reverberation time of the room.

He replaced Sabine's classical formula for reverberation time, $T = 0{\cdot}162 \times 10^{-2}\,V/A$, where V is the volume and A the total absorption of the room, by

$$T_2 = 0{\cdot}162 \times 10^{-2}(V + nV')/(A + nA') \tag{16}$$

where n is the number of resonators, and V' the effective increase in the volume of the room due to the presence of each resonator, which he called the "additional volume." The reverberation time is increased by the resonators if V'/A' is greater than V/A, and decreased if it is less.

The energy density in a room in which a steady sound field is maintained is given by $p_0^2/\rho c^2$, where p_0 is the r.m.s. pressure. At the resonance frequency of the resonator the r.m.s. pressure in the interior of the resonator will be Qp_0 and hence the energy density is $Q^2p_0^2/\rho c^2$. Assuming instantaneous energy exchange between room and resonator, a condition which Rschevkin shows to be satisfied in all practical cases, the resonator will contribute, from the viewpoint of energy storage, an effective additional room volume Q^2U. Hence, we may write

$$V' = Q^2U \tag{17}$$

For high values of Q and small values of μ therefore the increase of volume may result in a considerable increase in the reverberation time of the room.

Though Rschevkin's theoretical treatment for the individual resonator is sufficient to account for the main phenomena, experimental verification is not complete and the effects of interaction between neighbouring resonators are not considered. It was therefore found necessary to make tests to verify the main features of the theory, as it applies to single and combined resonators and to determine the practical limitations associated with the phenomenon.

3. EXPERIMENTAL

3.1. *Experiments on single resonators.*

The first resonators constructed were designed to have the ideal properties of rigidity and low internal loss. Six cylindrical tins each of 2 530 c.c. volume were cast into a concrete block, each tin communicating with the outer air through an opening 3·8 cm long and 4·1 cm in diameter. Sixteen of these blocks, shown in Fig. 2, were made, providing a total of 96 resonators. The calculated natural frequency of each resonator was 144 c/s and other frequencies could be obtained by inserting wooden bushes into the necks.

Two sets of bushes were made:

(*a*) $l_0 = 3{\cdot}8$ cm, $r = 1{\cdot}27$ cm, giving resonator frequency of 105 c/s;

(*b*) $l_0 = 1{\cdot}59$ cm, $r = 1{\cdot}27$ cm, giving resonator frequency of 75 c/s.

A wooden block of six resonators similar to (*b*) was also made.

Measurements of frequency and Q were first made on a single resonator, the others in the same block being corked. A small hole was bored in the back of the resonator cavity and a probe microphone was inserted, as shown in Fig. 1, through the hole into the cavity. The block of resonators was

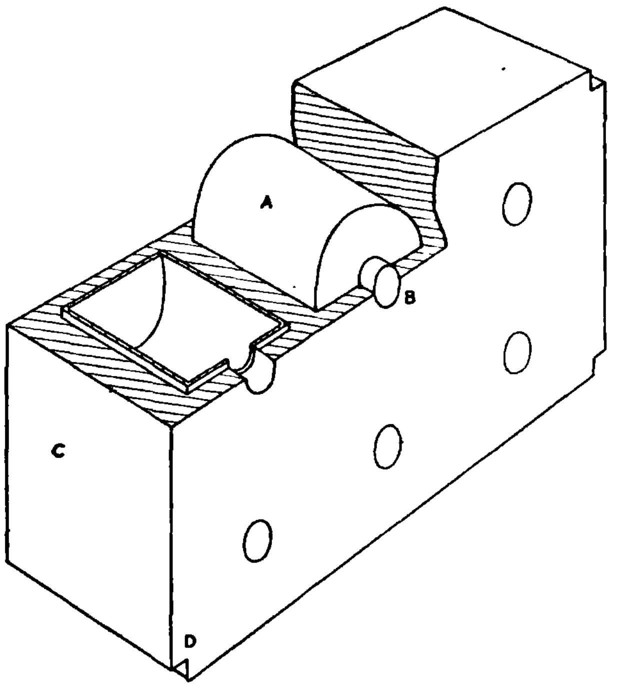


Fig. 2. Experimental block of resonators
A, metal cylinder; *B*, removable wooden bung; *C*, concrete body; *D*, hand grip.

Table 1. *Resistances of resonator necks*

	Q	R_2 *measured* (g sec⁻¹ cm⁻⁴)	R_2 *calculated* (g sec⁻¹ cm⁻⁴)	μ *(array)*	*Frequency* c/s
144 c/s resonator $\sigma = 225$ cm² $R_1 = 3 \cdot 7\times10^{-3}$ (g sec⁻¹ cm⁻⁴) (single resonator)					
Open neck	35–40	$1 \cdot 8\times10^{-2}$	$1 \cdot 08\times10^{-2}$	0·10	144
One layer bandage	15	4·5	—	0·24	144
Open mesh hessian	13	5·3	—	0·28	—
Two layers bandage	10	6·8	—	0·36	—
Cullum's scrim	10	6·8	—	0·36	—
Rockwool in cavity	9	7·6	—	0·41	—
Fibreglass fabric	8	8·5	—	0·46	140
Three layers bandage	5	13·6	—	0·73	—
Bandage in neck	5	13·6	—	0·73	—
105 c/s resonator $\sigma = 225$ cm² $R_1 = 1 \cdot 95\times10^{-3}$ (g sec⁻¹ cm⁻⁴) (single resonator)					
Open neck	18	$5 \cdot 4\times10^{-2}$	$3 \cdot 5\times10^{-2}$	0·29	—
One layer bandage	8	12·0	—	0·64	—
75 c/s resonator $\sigma = 225$ cm² $R_1 = 1 \cdot 28\times10^{-3}$ (g sec⁻¹ cm⁻⁴) (single resonator)					
Open neck	13	$9 \cdot 3\times10^{-2}$	$5 \cdot 1\times10^{-2}$	0·50	—
One layer bandage	6·5	18·5	—	1·00	—
Two layers bandage	5·5	22	—	1·18	—
Rockwool in cavity	5·5	22	—	1·18	—
Fibreglass fabric	3·5	34	—	1·82	—
Bandage in neck	3·5	34	—	1·82	—
75 c/s wooden resonator $\sigma = 295$ cm² ~~$R_1 = 1 \cdot 28\times10^{-3}$ (g sec⁻¹ cm⁻⁴)~~ (single resonator)					
Open neck	13	$9 \cdot 3\times10^{-2}$	$5 \cdot 1\times10^{-2}$	0·65	—
One layer bandage	6·5	18·5	—	1·30	—

placed in the open air and energized with tone of sl varying frequency from a loudspeaker 2 m away. L experiments were made indoors after it had been found the reflexions from the walls of the room did not affect results. The resonance curves obtained were used to calcu Q and hence, by equation (11), the total resistance $R_1 +$

The results are shown in Table 1. The value of R_2 sh in the second column is found by subtracting a calcul value of R_1 for an isolated resonator from the total resista $R_1 + R_2$ derived from Q. The third column shows, w appropriate, the value of R_2 calculated from equation (The measured values are seen to be somewhat higher tl the calculated values, possibly owing to eddy formati It will also be noticed from the table that the addition particular fabric across or within the neck raises the resista by a factor which is approximately the same for the 75 c/ for the 144 c/s necks. The resonance frequency was modified by the addition of the fabrics except where they t the form of wads stuffed into the necks. The fourth colu shows the value of μ as defined in Section 2.3, obtained fr R_2 in the second column and R_1 for an array as given equation (9), by division.

3.2. *Absorption measurements.*

Measurements were made in a reverberation cham having a volume of 28 m³ (1 000 ft³), using 84 individ resonators contained in 14 blocks. Control runs w made between experiments with the resonator holes bloc with corks. The same arrangement of resonators in chamber was maintained throughout the tests. The 144 resonators produced negligible absorption with open nec but with one layer of bandage over the holes a peak absorpti coefficient of 65% was reached at 140 c/s (Fig. 3).

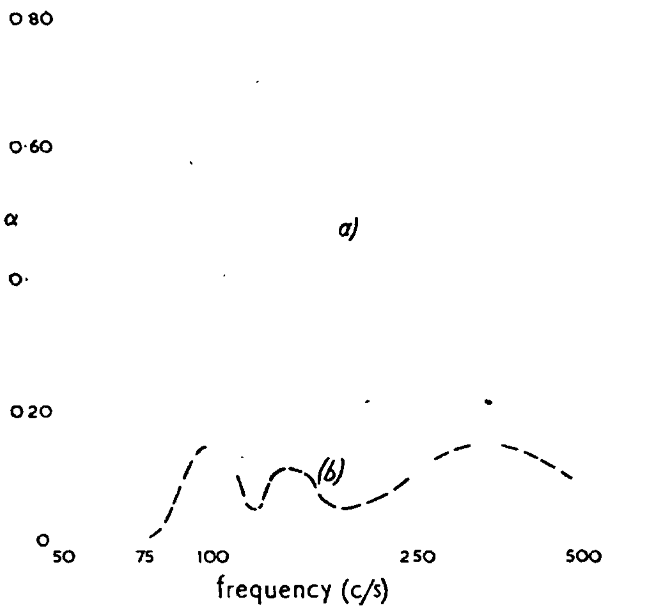


Fig. 3. Absorption coefficient of 144 c/s resonators
Curve (*a*) one layer of bandage, curve (*b*) open necks.

The subsequent work was done with the resonators fitte with the 75 c/s bushes, since these were more uniform i ~~diameter than the holes in the concrete castings. The 75 c resonators were tested with open necks,~~ with a wad of bandag inserted in each neck and with one and two layers of bandag across the orifice. The curves so obtained are shown in Fig.

eak of reverberation time at 75 c/s in the empty room e accurate measurements in this region difficult and many had to be made. The wads of bandage were observed ıry greatly in resistance and it was found preferable to use osely woven glass fabric where high resistances were ired.

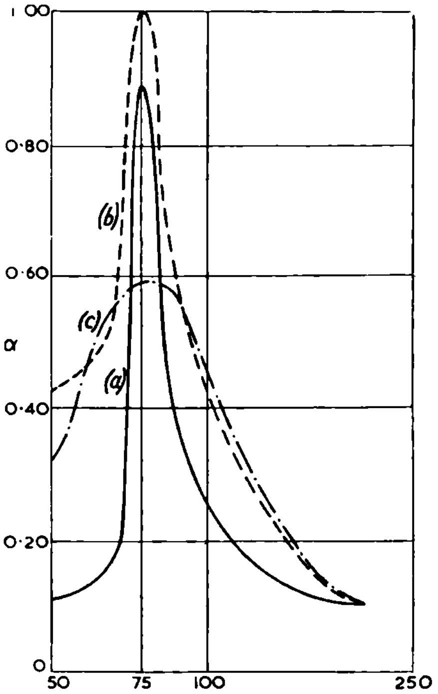
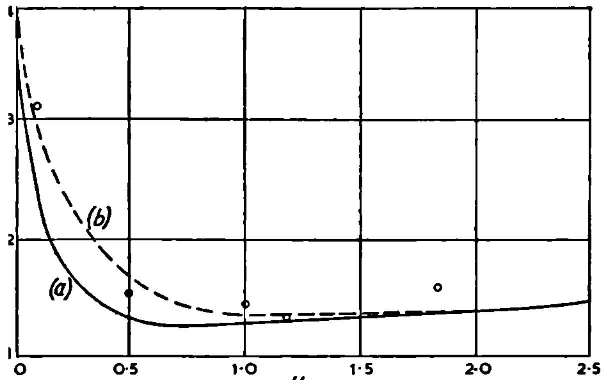


Fig. 4. Absorption coefficient of 75 c/s resonators
Curve (*a*) open necks, curve (*b*) two layers of bandage, curve (*c*) wad of bandage in neck.

he results of the measurements at 75 c/s are plotted in 5 for comparison with the theoretical curves. The nate of each point represents the reverberation time at nance frequency, while the abscissa shows the value of μ ved from Q measurements. The lowest time was obtained ı a resistance corresponding to a μ of 1·18 (two layers of dage), but the optimum value was not critical.



5. Calculated and measured reverberation times in tiled room
›erimental points, curve (*a*) simple theory, curve (*b*) corrected curve allowing for energy storage.

ıble 2 shows a comparison between absorption coefficients ved from the measurements by means of equation (15) and Eyring's formula for reverberation, disregarding the effect of additional volume. The agreement is satisfactory except in the case of the 144 c/s resonators with open necks, where the measured coefficient is very much too low, suggesting that energy storage in the resonators may be having an appreciable effect. The magnitude of this effect is consistent with this view, the calculated "extra volume" from equation (17) being 37 m³ as compared with 28 m³, the volume of the reverberation room. If the real absorption coefficient is taken as 0·33, from equation (15), the apparent coefficient from Eyring's formula would be $0 \cdot 33 \times 28/(28 + 37)$ approx. $-$ 0·14 approx.

The corrections for the other cases in the table are negligible.

Table 2. *Absorption coefficient of resonators in array*

Resonator	μ	*Absorption calculated*	*coefficient measured*
144 c s open neck	0·10	0·33	0·14
144 c/s one layer bandage	0·24	0·62	0·65
75 c/s open neck	0·50	0·90	0·88
75 c/s one layer bandage	1·00	1·00	0·92
75 c/s two layers bandage	1·18	0·99	0·99
75 c/s glasswool fabric	1·82	0·90	0·75

3.3. *Measurements in studios.*

The abnormal conditions of the reverberation room were very suitable for the measurement of absorption, since the ratio of volume to absorption was high and the increase of absorption due to the resonators would be expected to outweigh any possible increase in effective room volume. This condition would not hold in any ordinary room in which the average absorption coefficient of the walls might be as high as 0·2 or more, and accordingly tests were made in three such rooms, one quality listening room and two small talks studios. The same 84 resonators were used, tests being made with holes open, closed and covered with resistive materials.

The listening room had a reverberation time of approximately 0·65 sec at 75 c/s with the resonators blocked. This was reduced by the action of the resonators whether the holes were open, covered with fabric or filled with bandage, corresponding to a range of μ from 0·55 to 2·0. The reductions of reverberation time were in accordance with the values of μ.

The first talks studio had a volume of 43 m³ (1 500 ft³) and a reverberation time of 0·35 sec at 75 c/s, falling to 0·32 sec at 144 c/s. This was treated with 75 c/s resonators open and matched to give $\mu = 1$, with blocked resonators as a control, and 144 c/s resonators with open necks. The 144 c/s open resonators produced a very marked reverberation time increase from 0·32 sec to 0·52 sec at 144 c/s which, by the use of Table 2 and equation (16), may be shown to correspond to an "additional volume" of 28 m³ (1 000 ft³). Traces obtained with a logarithmic high-speed level recorder were linear, indicating that there was rapid exchange of energy between the resonators and the room. Careful listening tests using short pulses of tone, moreover, revealed this peak of reverberation but no tendency for the pitch of neighbouring tones to be modified during decay.

Another striking observation was that the resonators did not appear to "colour" speech from the studio heard over a microphone-loudspeaker listening chain. The increase of reverberation time at 144 c/s corresponds to an additional volume of 27·5 m³ (970 ft³).

The second talks studio was an experimental studio of approximately the same volume as the first with a reverberation time of 0·47 sec at 144 c/s. Fig. 6 shows the effect of 84 resonators with open necks (*b*) and with two layers of bandage (*c*), both tuned to 144 c s. The increase of reverberation time in (*b*) corresponds to an additional room volume of 22·5 m³ (790 ft³), a figure close to that obtained in the first studio. As in the first studio this pronounced peak of reverberation had very little effect on the subjective qualities of the room as a talks studio. Indeed, disk recordings made in the studio with the resonator necks open were actually preferred by a majority of observers taking part in controlled listening tests to those made with matched or closed resonators. There was no audible coloration at 144 c/s though it may be that the effect, if present, was insignificant in comparison with a previously existing coloration at 115 c/s due to a structural resonance.

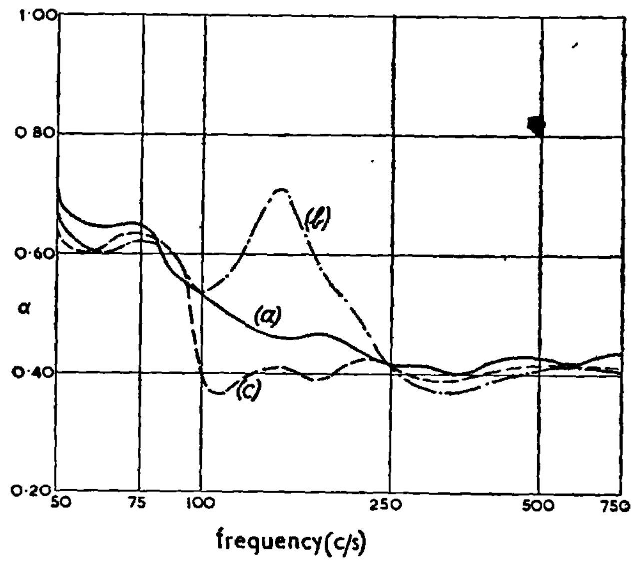


Fig. 6. Reverberation time measurements in second studio
Curve (*a*) necks closed, curve (*b*) necks open $\mu = 0 \cdot 07$, curve (*c*) necks with additional resistance $\mu = 0 \cdot 45$.

Following these observations some subjective tests were made on the influence of resonators of various frequencies in determining speech quality. The tests were made in a dead room and gave differing results according to the spectrum of the voice used.

4. DISCUSSION OF RESULTS

4.1. *Frequency of resonance.*

It was found that measurements of resonance frequency were in exact agreement with calculation. The concrete castings designed for 144 c/s, for example, gave a mean measured figure of 144 c/s with a standard deviation of ± 3·8 c/s for individual resonators. This variation corresponds to the observed variation in neck diameter. The 75 c/s bushes gave a smaller standard deviation on account of their greater uniformity. Prediction is therefore sufficiently accurate since these variations are considerably smaller than the bandwidths normally used.

4.2. *Absorption.*

It is not possible to predict the amount of absorption so accurately, since it depends on the nature of the sound-field and the arrangement of the resonators. Table 2 shows a remarkable agreement between calculation and measurement for arrays, the measurements having been carried out in a reverberation room in which, at low frequencies, the sound field consists of plane waves. The values of additional volume quoted in Section 3.3 above, however, imply values of Q, after correction for frequency variation between individual resonators, from 41 to 49, of the same order as for an open-necked resonator in a hemispherical sound rather than in a plane-wave field. Further experimen elucidate this showed that in the reverberation chambe radiation resistance was substantially unaffected by arra ment. Moreover, if the resonators were tuned to the quency of the second vertical mode of the room and pl upon the floor, absorption reached a maximum if the resistance corresponded to the plane-wave resistance, whe the resonators were grouped together or separated.

It is therefore clear that the performance of a reson depends upon the nature of the sound field in the room very considerable degree, and it is necessary to study room modes about the resonance frequency before deci upon the position of the resonator. Alternatively, all prac designs should make provision for varying the neck resista

The low Q values of arrayed resonators are in appa contradiction to the high values of resistance required maximum absorption. The latter requirement is in a with other workers,[5, 6] who find that there is good between the absorption of arrays consisting of perfor panels and the resistance figures measured by the impeda tube method. There are, however, no published values of resistance of open necks measured when arranged as array ordinary room conditions.

The apparent contradiction arises from the interact between neighbouring resonators. When the driving sou field ceases these tend to get into antiphase and act toget as dipoles or more complicated combinations in which resistance presented to each resonator is low compared that of the room. The individual variations in resona frequency facilitate this change of regime. The measu Q values for open neck resonators is therefore low. When neck resistance is artificially raised to achieve maxim absorption of the sound in the room, the resistance presen to each resonator by the neighbouring necks becomes at le as high as the $\rho c/\sigma$ resistance of a plane-wave field, and tendency to interact is greatly reduced. Maximum absorpti is therefore obtained with resistance values suitable for pla wave conditions. This view has been checked qualitativ by experiments using a large number of bottles which co be arranged in any manner in either the reverberation ro or an acoustically treated studio.

The conclusion is that if accurate prediction is desired choice must be made between isolated resonators and co paratively large plane arrays in which the value of μ is un or greater. The question of bandwidth has not been sidered in detail as this has been treated fully by Kosten a Zwikker.[5] The bandwidth of an isolated resonator w $\mu = 1$ is extremely small, and these authors recomme μ values between 10 and 40 in order to obtain a suffici width, thereby reducing the peak absorption to a sm fraction. The resistance of an array, being high in co parison with its inertance, gives an absorption band reasonable width even at values of μ approaching unity, a this width may be increased with only a second-or reduction in peak absorption.

5. PRACTICAL FORMS OF HELMHOLTZ RESONATOR

The experimental resonators described above were design for the special purposes of quantitative investigation. Whe manufacture in large quantities is intended, some simplific tion is required. Other shapes may be used for architectur reasons, and, to a certain extent, existing parts of the buil may be used as the cavities. Simplification and incr ncy may be effected by mounting the resonators in a array, with groups of equally spaced holes having ıon air spaces. It is desirable in this case not to allow ıunicating air spaces to extend more than a fraction of 'elength and, therefore, partitions must be provided at ›le intervals, say every 2–3 ft, in a low-frequency absorbray. Stiffening members would in any event be required ervals of this order.

ınd of low frequency may be absorbed by resonators of ımall size and depth compared with other types of low ·ncy absorber. For instance, the volume of the experi- ıl 75 c/s resonators could have been reduced to one-tenth ı suitable adjustment to the neck diameter. A reduction : can only be made, however, by reducing the width of requency band over which effective absorption takes so that the advantages of small dimensions are outed by the necessity to provide a greater number to b over a given range of frequencies. Herein lies the flexibility of the method, since almost any basic dimen- may be chosen to fit the architectural design of the ng.

; possible also to use the resonator units as sound diffusers. ·imental cylindrical diffusers have been made from ıs plaster and mounted so that the slit between the edge ch resonator and the wall, in conjunction with the v interior, formed a resonator of frequency 80 c/s. ›ximate agreement was obtained between the frequency he theoretical value for a slot resonator given by Peder- › and with a resistive material in the slot, broad or ive absorption was obtained, the absorption coefficient : latter case having maximum values up to 100% based ıe area of wall covered by the diffuser. Rectangular ers or coffering could be used in a similar manner, such s also being preferable for other reasons (see Somerville Vard[8]).

ıonators arranged in rows, having properties intermediate en the isolated resonators and two-dimensional arrays, end themselves readily to architectural designs and to :tion. They are being used extensively for a music › now under construction by the B.B.C.

ference should also be made to perforated panel ators for middle and high frequency absorption. These been treated very completely by several authors, notably slev[6] and Kosten and Zwikker.[5] The range of absorption may also be extended by adding a layer of rockwool and a perforated cover over the front of an array of low frequency resonators, the rockwool being adjusted in density and thickness to provide the necessary matching of the resonators and to give an overall curve of the required shape.

6. CONCLUSIONS

The experimental work described in this paper establishes that Helmholtz resonators tuned to low frequencies form an extremely flexible method of absorbing sound. The most effective way of using them is to arrange a comparatively large number as a series of plane arrays and to raise the neck resistance to a value rather higher than the radiation resistance. Isolated resonators are capable of very large absorption, but this cannot be used, because this condition results in too small a bandwidth for practical purposes, and involves the possibility of coloration due to re-radiation. The frequency of maximum absorption may be predicted accurately, but it is wise in designing studio treatments to allow for adjustment of the neck resistance of the resonators, since this is very much simpler than trying to make allowance for the pressure distribution in the room over the absorption band. This last conclusion applies equally to other absorbers.

7. ACKNOWLEDGMENTS

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MEMBRANE SOUND ABSORBERS AND THEIR APPLICATION TO BROADCASTING STUDIOS

By C. L. S. Gilford, M.Sc., F.Inst.P., Research Department, B.B.C. Engineering Division.

INTRODUCTION

TO provide means for the absorption of sound at medium and high frequencies is a comparatively simple matter and in most applications of acoustic treatment all that is required. For the reduction of noise in restaurants and offices, for example, it is unnecessary to absorb sound of very low frequencies as it contributes little to the subjective impression of noise. Similarly if it is desired to improve the acoustics of conference rooms or small assembly halls, attention may be directed principally to the upper and middle frequencies which mainly determine intelligibility, while the vibration of walls, ceilings, and wood panelling, and the transmission through doors and windows, usually provide enough lower-frequency absorption. For this reason commercially available sound absorbers consist invariably of porous materials which are extremely efficient over most of the audio-frequency range, although ineffective below about 200 c/s.

Low-frequency absorption cannot, however, be ignored in the treatment of concert halls or of broadcasting and recording studios. In the former, for example, excessive bass reverberation can cause masking of the lighter instruments by loud instruments of low frequency. Many of the older concert halls, however, have fortunately escaped this fault by the extensive use of wood panelling which absorbs effectively in the bass by frictional losses in the panels when they are caused to vibrate by the sound pressure. In the case of broadcasting or recording studios a further consideration enters because the brain, being deprived of the directional information from its normal binaural system, is unable to discriminate against the reverberant sound in favour of the direct sound. For example, a room which may be perfectly pleasant to converse in is usually useless as a talks studio; it will probably sound too reverberant, and faults due to inadequate absorption at low frequencies will be particularly noticeable.

Panels of plywood or fibreboard, as frequently used for bass absorption in broadcasting studios, have several disadvantages. Different samples of similar panels may vary considerably in their absorption which is also affected by the way in which the edges are fixed. Unwanted acoustic effects may also be caused by re-radiation at a resonance frequency, and although variation of the individual frequencies would mitigate the effect, the addition of weights or variation of the spacings of the fixings usually fails to give a worth-while diversity.

During the war, C. W. Goyder* used linoleum panels for selective low-frequency absorption in broadcasting studios at New Delhi. Since the rigidity of linoleum is small, the frequency of maximum absorption is determined mainly by the mass per unit area of the panel and the depth of the air space enclosed behind it. The system may therefore be designed to absorb at any desired frequency within practical limits with reasonable uniformity of behaviour. This article describes subsequent developments which have been carried out by the B.B.C. Engineering Research Department, leading to the successful use of flexible membrane absorbers in studios of all sizes.

* Goyder, C. W. Private communication 1943

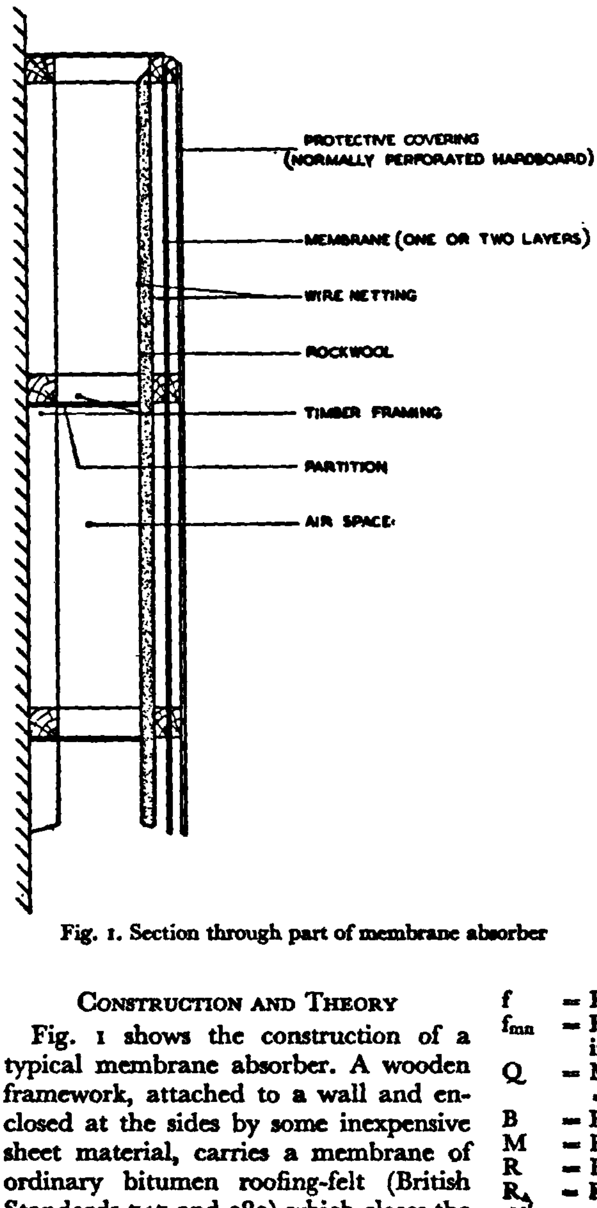

Fig. 1. Section through part of membrane absorber

Construction and Theory

Fig. 1 shows the construction of a typical membrane absorber. A wooden framework, attached to a wall and enclosed at the sides by some inexpensive sheet material, carries a membrane of ordinary bitumen roofing-felt (British Standards 747 and 989) which closes the front of the framework. A blanket of a loose fibrous absorber, such as Cabot's quilting, glass fibre, or rockwool supported by wire netting, is sometimes hung behind the membrane for reasons which will be made clear later, and in front of the membrane is fixed a protective cover which may also serve as an acoustic filter* to modify the absorption-frequency characteristic.

The air space enclosed behind the membrane is divided up by partitions which serve to prevent lateral transverse air motion at the operating frequencies. The main restoring force when the membrane is deflected is due to the elasticity of the enclosed air. For the purpose of calculating the resonance frequency it will be assumed that the general shape of the membrane is similar to that of a stretched elastic membrane in which the restoring force is due to elastic tensions.

The following notation will be used:

A	= Area of membrane
σ	= Mass of membrane per unit area
V	= Volume of air space in equilibrium position
l	= Mean depth of air space
p	= Atmospheric pressure
c	= Velocity of sound in air
ρ	= Density of air
γ	= Ratio of specific heats of air
a, b	= Lengths of sides of rectangular membrane
f	= Frequency at resonance of membrane
f_{mn}	= Frequency at resonance of membrane in (m,n) mode
Q	= Magnification of resonant system = $\omega M/(R + R_A)$
B	= Bandwidth of resonant system
M	= Effective mass of membrane
R	= Effective resistance of membrane
R_A	= Radiation resistance of membrane.
M'	= Acoustic inertance of membrane. = $\dfrac{M}{(ab)^2}$

Resonance Frequency

Consider a rectangular membrane bounded by the lines $x = 0$, $x = a$, $y = 0$, $y = b$, and an elementary area dxdy of the membrane surrounding a general point (x, y).

* Bolt, R. H. *Journal of Acoustical Society of America*, Vol. 19, p. 917 (1947)

If Z (x,y) is the deflection at (x,y), Morse* gives:

$$Z(x,y) = h(t)\sin\frac{m\pi x}{a}\sin\frac{n\pi y}{b}$$

where m, n are integers, h(t) is the peak deflection ......(1)

The kinetic energy of the whole membrane is therefore

$$\tfrac{1}{2}\sigma\int_0^a\int_0^b\left\{\frac{dZ(x,y)}{dt}\right\}^2 dxdy = \sigma\left(\frac{dh}{dt}\right)^2\frac{ab}{8} \quad \text{......(2)}$$

If we ignore elastic restoring forces entirely, the potential energy of the system is altered only by changes in the volume of the air in the cavity. The excess pressure in the cavity due to a reduction in volume δV is $\gamma p\delta V/V$ and, assuming small oscillations, the mean pressure during deflection from equilibrium position will be $\tfrac{1}{2}\gamma p\delta V/V$, acting everywhere perpendicular to the membrane. Hence the potential energy at deflection Z is given by $\tfrac{1}{2}\gamma p(\delta V)^2/V =$

$$\tfrac{1}{2}\gamma p/V\left\{\int_0^a\int_0^b Z(x,y)\,dxdy\right\}^2 \quad \text{..... (3)}$$

$$= \begin{cases} \dfrac{8\gamma ph^2}{V}\cdot\dfrac{a^2b^2}{\pi^4n^2m^2}\ ; & \text{m, n both odd} \\ 0 & ;\ \text{m, n not both odd.} \end{cases}$$

Since the sum of the kinetic and potential energies is conserved, neglecting losses,

$$\begin{cases} \dfrac{\sigma ab}{8}\left(\dfrac{dh}{dt}\right)^2 + \dfrac{8a^2b^2\gamma ph^2}{\pi^4Vn^2m^2} = C_1; & \text{m, n both odd} \\ \dfrac{\sigma ab}{8}\left(\dfrac{dh}{dt}\right)^2 = C_2; & \text{m, n not both odd} \end{cases}$$

where C_1, C_2 are constants.

Differentiating with respect to t and dividing by $2\dfrac{dh}{dt}ab$,

$$\begin{cases} \dfrac{\sigma}{8}\dfrac{d^2h}{dt^2} + \dfrac{8\gamma pab}{\pi^4Vn^2m^2}\cdot h = 0 & \text{m, n both odd} \\ \dfrac{\sigma}{8}\dfrac{d^2h}{dt^2} = 0 & \text{m, n not both odd.} \end{cases}$$

The first of these two equations will be recognized as representing simple harmonic oscillations of frequency given by:

$$f_{mn} = \frac{4}{\pi^2 mn}\sqrt{\frac{\gamma pab}{\sigma V}};\ \text{m, n, both odd} \quad \text{......(4)}$$

$$= \frac{153}{mn\sqrt{\sigma l}} \text{ in c.g.s. units,}$$

since $\gamma = 1{\cdot}4$, $p = 10^6$ dynes/cm² at N.T.P., and $V/ab = l$, the depth of the air space.

If m, n, are not both odd, there is no oscillation.

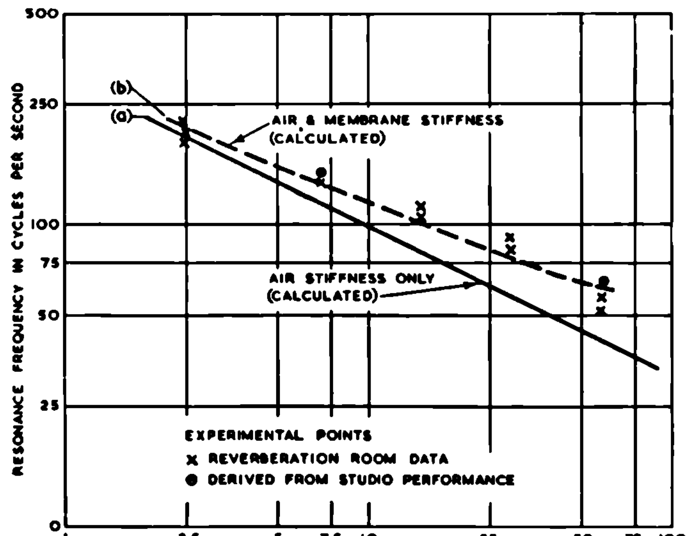

Fig. 2. Relationship between air space depth and frequency: (*a*) calculated, assuming air stiffness only; (*b*) calculated, assuming air and membrane stiffness

Fig. 2 shows this relationship for the simplest mode of vibration where m = 1, n = 1, with experimental points for comparison. The membrane material is roofing-felt of mass 0·236 g/cm². It is clear that the simple calculation gives too low a frequency, particularly for the deeper units, the

* Morse, P. M. *Vibration and Sound*, 1st edn. p. 142 (New York: McGraw Hill Book Co., 1936)

error being comparatively small for the shallowest. This behaviour may be explained most readily by the assumption of a constant elastic restoring-force term which is independent of the depth of the unit, depending only on the physical properties of the membrane itself. In the case of the deepest units the restoring force due to the air pressure is small and of the same order as that due to membrane stiffness, but in the shallowest units the air pressure restoring force is very much greater and membrane stiffness is almost negligible.

The dotted line of Fig. 2 shows the variation of frequency with depth, assuming an arbitrary membrane stiffness. The value chosen, equal to the air stiffness for a depth of 47 cm., gives the best fit to the experimental points. The agreement is good enough to confirm the approximate theory given above and therefore gives a useful indication of the relative magnitudes of the two types of restoring force. Consideration of Fig. 2 shows that the two forces are equal in an absorber tuned to 66 c/s; at any other resonance frequency f, the ratio between them is therefore given by:

$$\frac{\text{Air stiffness}}{\text{Membrane stiffness}} = \left(\frac{f}{66}\right)^2.$$

A further consequence follows from the theory; the highest frequency of resonance occurs when all parts of the membrane move in the same direction at the same time, i.e., when $m = n = 1$. Even modes are absent and odd modes such as (1, 3) are of lower frequency, even having regard to membrane-elastic forces, and are not easily excited if the membrane is small compared with a wavelength at the (1, 1) resonance frequency.

In this respect this type of structure differs profoundly from other resonant absorbers such as wooden panels, which oscillate readily in complex modes at frequencies higher than the fundamental. The absence of effective partials may account partly for the fact, dealt with more fully below, that properly designed membrane absorbers do not 'colour' the sound in a studio.

Theory of the Absorption

Expressions for the absorption of sound by bodies small in comparison with the wavelength have been given by several authors, but in most cases the theory given is confused or actually wrong. If W is the r.m.s. volume flow of air in the immediate neighbourhood of the absorber, i.e., of the membrane in this instance, and R its effective internal acoustic resistance, the power absorbed will be given by RW^2. The value of W depends upon R, R_A and the sound pressure and it is here that confusion has arisen.

Thévenin's network theorem states that 'the current in a terminating network of impedance Z connected to any network is the same as if Z were connected to a generator whose voltage is the open-circuit voltage of the network and whose internal impedance Z_A is the impedance looking back from the terminals of Z, with all generators replaced by impedances equal to the internal impedances of these generators'. The open-circuit voltage is equivalent to the sound-pressure which would result if the resonator were replaced by a perfectly reflecting wall, i.e. 2p. The impedance looking back is the radiation impedance and hence the volume current W through the resonator at the resonance frequency is $2p/(R + R_A)$, all reactive components of the impedances being cancelled at resonance, so $W = 2p/R_A(1 + \mu)$ where $\mu = R/R_A$.

This result is perfectly general, and the presence of the factor 2 does not depend upon the physical existence of a hard surface surrounding the resonator, as has been suggested elsewhere. It will also appear, for example, in the same calculation applied to one semi-infinite tube terminating another. The rate of energy absorption is therefore

$$W^2R = 4p^2\mu/R_A(1+\mu)^2 \qquad \text{—(5)}$$

This is equal to the energy in an area G of a plane wave, sound pressure p if

$$p^2G/\rho c = 4p^2\mu/R_A(1+\mu)^2$$

$$\textit{i.e.}\ \ G = 4\mu\rho c/R_A(1+\mu)^2 \qquad \text{—(6)}$$

The maximum effect is obtained if $\mu = 1$ and hence

$$G = \rho c/R_A \qquad \text{.....(7)}$$

On either side of the resonance frequency the volume current and therefore the absorption falls, the shape of the absorption-frequency curve being calculable from the well-known properties of resonant systems (see for example Terman*). The velocity response of the membrane falls to $1/\sqrt{2}$ of its resonance value at frequencies differing from f by $\pm f/2Q$ where Q is the magnification, given by

$$Q = 2\pi f M/(R+R_A) \qquad \text{.....(8)}$$

In the same frequency interval the absorption falls to $\frac{1}{2}$ of its peak value.

The bandwidth B will be defined as the frequency interval given by

$$B = f/Q \qquad \text{.....(9)}$$

The optimum condition for absorption is obtained if $R = R_A$. Increase of R above this value results in reduced absorption but increases the frequency band over which effective absorption occurs. In using absorbers of this type for a general reduction of reverberation time over a wide range of frequency it is therefore an advantage for R to exceed R_A rather than the reverse.

In the frequency range 50–300 c/s for which the membrane absorbers are most useful, the linear dimensions are smaller than a wavelength. The radiation resistance therefore changes considerably with size and frequency.

Crandall† has shown that at frequency f the radiation resistance of a piston of radius r forming part of an infinite wall is given by

$$R_A = \frac{\rho c}{\pi r^2}\left\{1 - \frac{J_1(2kr)}{kr}\right\} \qquad \text{.....(10)}$$

where $k = \frac{2\pi f}{c}$ and J_1 is the Bessel Function of the first kind of order 1.

The values of this expression are plotted in Fig. 3. Given experimentally determined values of the internal resistance R of the membrane it is possible by means of this figure to examine the behaviour of a unit of any desired size and frequency of resonance. Further discussion will be given below.

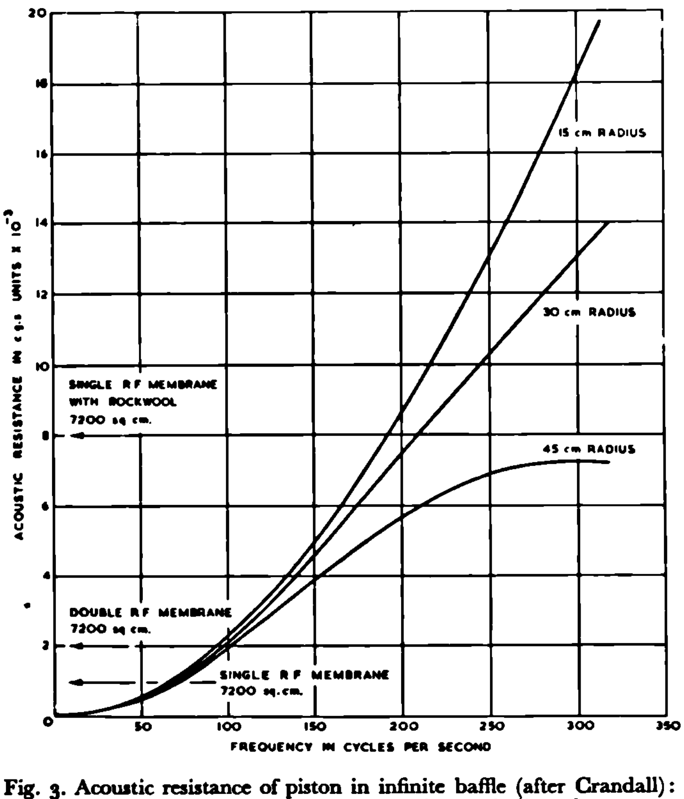

Fig. 3. Acoustic resistance of piston in infinite baffle (after Crandall): typical values for roofing felt membranes indicated

Experimental Work

Experiments on membrane absorbers comprised (1) a study of the vibration of

* Terman, F. E. *Fundamentals of Radio*, pp. 27–38 (New York: McGraw Hill Book Co., 1938)

† Crandall, I B. *Vibrating Systems and Sound*, p. 146 (New York: Van Nostrand Co., 1926)

single membranes mounted on a strong brick box, and (2) absorption measurements on larger numbers by the reverberation method.

(1) *Single Membranes*

Two brick structures were built. The first was used for measurements on membranes with an enclosed air space behind; it consisted of a brick box built on a concrete floor, the membrane to be tested being clamped across the open top which had inside dimensions of 50 cm. × 50 cm. The second, used for measurements on membranes not enclosing an air space,

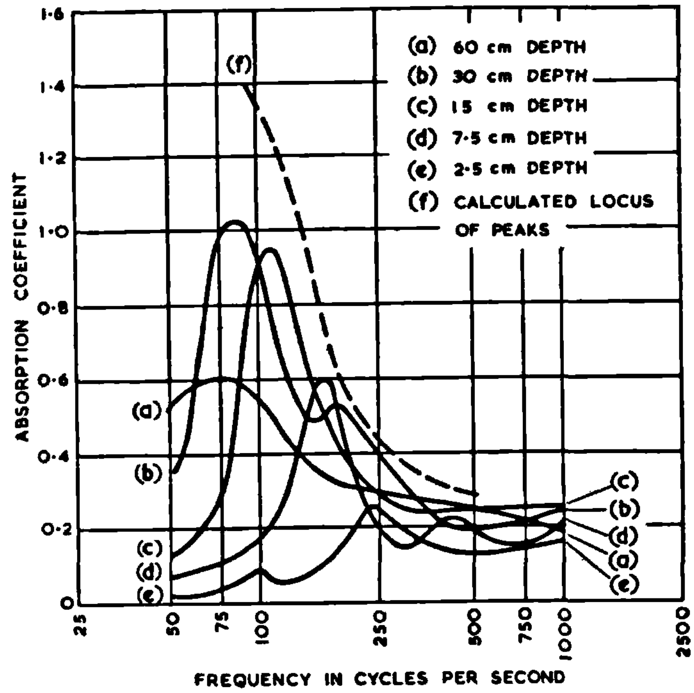

Fig. 4. Absorption coefficient of single membranes without rockwool

consisted of an open vertical square frame of the same size, across which the membrane could be clamped.

The vibration of the membrane was measured by means of a velocity pick-up, the output being applied to a cathode-ray oscilloscope with a single-sweep time base. The membrane was excited by a sharp tap and the oscilloscope time base started simultaneously. The damped trains of waves were then photographed in order that the frequency and decrement could be measured. The results of these measurements were inconclusive, mainly due to the large damping in most of the materials, but some interesting observations were made. The oscillograms obtained with very flexible materials such as roofing-felt and thin polyvinyl chloride showed sinusoidal uniformly damped wave trains. Rigid materials, such as hardboard and asbestos millboard, gave very irregular decays suggesting the co-existence of several modes. Materials such as linoleum and fabric-backed polyvinyl chloride gave less irregular results.

(2) *Absorption Measurements*

The absorption measurements were made in a reverberation room by measurement of the reverberation time with and without the samples. A total absorbing area of 10 m^2 was used for each test, the membranes being mounted on wooden boxes of frontal areas 60 cm. square or 60 cm. × 120 cm. Air space depths ranged from 2·5 to 60 cm. The absorption due to the sides of the boxes was determined by separate experiments and subtracted from the total measured absorption of the complete units. Eyring's* formula was used for calculating the total absorption from the measured reverberation time, all results being given as a coefficient obtained by dividing the total absorption by the membrane area.

Comparisons were made between different numbers and groupings of the boxes.

* Eyring, C. F. *Journal of Acoustical Society of America*, Vol. 1, p. 217 (1930)

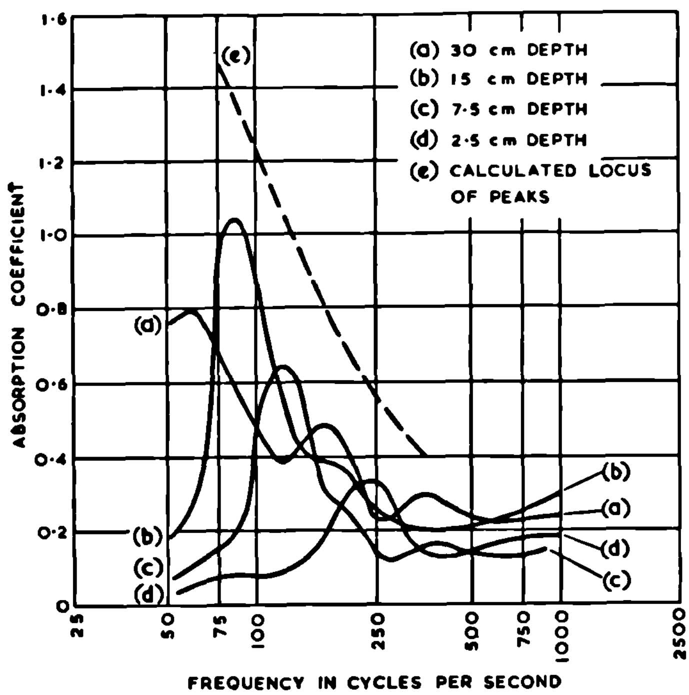

Fig. 5. Absorption coefficient of double membranes without rockwool

There was no apparent variation in absorption coefficient with the number of units but absorption was some 5 per cent. greater on the average with the units arranged singly than when placed in small groups. The results quoted in this article are for small groups distributed around three of the walls and on the floor. All membranes were covered with perforated hardboard fronts the action of which is to reduce the absorption at high frequencies by the rough surface of the membrane.

Fig. 4 shows the measured absorption characteristics of membranes of mass 0·236 grams/cm. over air spaces up to 60 cm. in depth. The curves of Fig. 5 are for units in which the membranes consisted of two layers of roofing-felt in contact with each other. Fig. 6 shows the results for single membranes backed by 5 cm. blankets of rockwool, the rockwool being held clear of the membrane. In the 2·5 cm. deep absorbers the whole air space was packed with rockwool.

The most striking feature of the families of curves shown in Figs. 4 and 5 is the reduction of absorption as the resonance frequency rises, rendering the shallower units relatively ineffective. When the membrane is backed by a rockwool blanket to increase the internal resistance (Fig. 6) this rapid reduction does not take place.

An explanation is found by reference to the radiation resistance curves of Fig. 3. In order to calculate the absorption it is also necessary to know the internal resistance R of the membrane. This is calculated in the following manner. The absorption bandwidth is used to derive the value of Q at the resonance frequency according to equation (9) whence, from a knowledge of the value of M, the effective mass of the membrane, the total resistance $R + R_A$ can be calculated from equation (8).

If the membrane is assumed to be vibrating to equation (1) we may calculate an 'effective mass' for the membrane, defined as that mass which, having a displacement and velocity equal to that of the point of greatest deflection, also has an equal kinetic energy.

From equation (2)

$$\tfrac{1}{2}M\left(\frac{dh}{dt}\right)^2 = \sigma\left(\frac{dh}{dt}\right)^2\frac{ab}{8}$$

and therefore $M = \dfrac{\sigma ab}{4}$ whence $M' = \dfrac{\sigma}{4ab}$

Inserting this value in equations (8) and (9) we get

$$R + R_A = \frac{2\pi f M'}{Q} = 2\pi M' B$$

Therefore $R = \dfrac{\pi\sigma B}{2ab} - R_A$ ....(11)

The values of R thus obtained are only approximate, owing to the large bandwidth. The average value is about 10^{-3} c.g.s. unit for single membranes, $2 \cdot 10^{-3}$ for double membranes, and $8 \cdot 10^{-3}$ for single rockwool-backed membranes. These figures are shown in Fig. 3 for comparison with the R_A curve for the 45 cm. radius circle which has the same area as the membranes used. The dotted curves on Figs. 4–6 are the calculated loci of absorption peaks for the three types of membrane using the experimental values of R and equation (6). The agreement with the experimental peaks is thought to be good enough to verify the consistency of the theory, demonstrating as it does the increase of performance of the higher-frequency units with the addition of rockwool. The failure to reach the high theoretical absorption coefficients below 100 c/s is probably due to the lack of diffusion in the reverberation room, which is of too small a volume for reliable determinations at such low frequencies.

Application of Membrane Absorbers to Studio Treatment

The requirements of a low-frequency absorber for use in studios are as follows:

(1) Absorption must be as high as possible over the frequency range desired.

(2) The bandwidth as defined in equation (9) should not greatly exceed one half-octave, since the total existing absorption in a studio may vary rapidly with frequency, and narrow-band absorbers are required for correction of this fault.

(3) It must not re-radiate sound appreciably over any narrow frequency band since this causes colourations.

(4) It must be adaptable for use in studios with reverberation times from 0·2 secs up to nearly 2 secs.

(5) It must be cheap and easy to construct by ordinary building methods.

All these requirements are adequately satisfied by the membrane absorber using roofing-felt. The peak absorption is very high and the bandwidth is reasonably small. It will now be shown, however, that the lower limit of bandwidth is determined by the third requirement, absence of colouration.

Consider the behaviour of a resonant absorber excited by a short pulse of sound such as a speech syllable. During the pulse the amplitude of the resonator will increase towards its steady-state value, energy flowing from the room into the resonator. This flow continues after the end of the pulse if the steady-state condition has not been reached, and the resonator continues to absorb. If the rate of decay of the resonator vibrations is less than that of the sound-field, however, a time will be reached when the energy flow

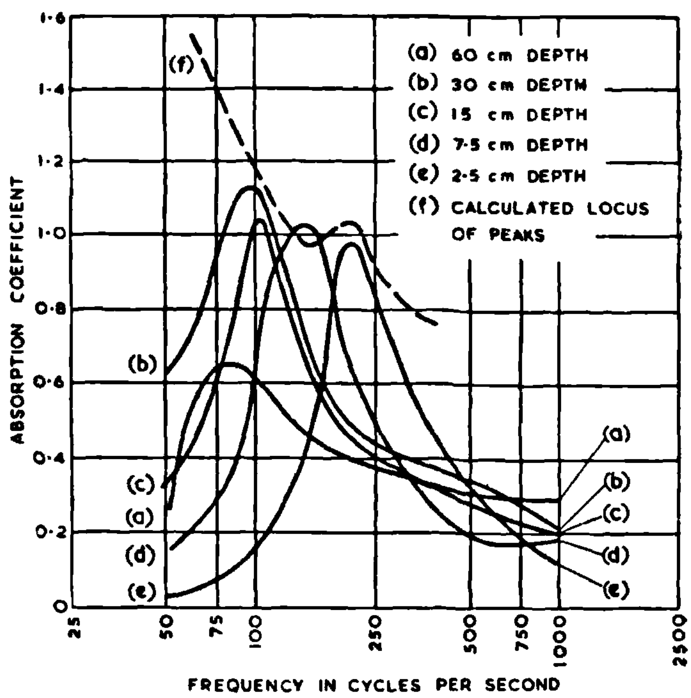

Fig. 6. Absorption coefficient of single membranes with rockwool backing

becomes reversed, and the resonator behaves thereafter as a single-frequency sound source.

If the amplitude of vibration at the transition point is A, at time t it will be

$$A \exp\left\{-(R+R_A)t/2M\right\} \quad \text{---(12)}$$

$$= A \exp(-\pi ft/Q) \quad \text{---(13)}$$

The amplitude will have fallen to A/1000 after a certain time T given by $T = (\log_e 1000)\, Q/\pi f$.

But by equation (5), $Q = f/B$

Hence $T = (\log_e 1000)/\pi B = 2\cdot 2/B$ ---(14)

This general result applies to any resonant absorber whatever and therefore includes Helmholtz resonators, wood panels etc. The time T is known as the 'decay time' of the resonator by analogy with the reverberation time, similarly defined, of a room.

Experience has shown that to avoid colouration the decay time must be appreciably smaller than the reverberation time of the room. This therefore defines the lower limit of possible bandwidths. For a small talks studio the reverberation time may be approximately 0·25 secs and the decay time of any absorber should be less than, say, 0·15 secs. This places a lower limit of $\frac{2\cdot 2}{0\cdot 15} = 15$ c/s, on the bandwidth. A glance at the absorption curves of Figs. 4 and 5 shows that the bandwidth even for the membranes without rockwool blankets is greater than this at the lowest frequencies, increasing with the frequency. The rockwool-backed units have a larger bandwidth.

The fact that any absorber with a bandwidth less than 15 c/s will cause colourations in a small studio explains the failure of many past attempts to damp out troublesome structural or standing-wave resonances by selective absorption. The colourations arising from such causes are highly frequency-selective, having bandwidths of a few cycles only. Narrow bandwidth absorbers will themselves reradiate, and wider-bandwidth absorbers cause a general reduction of reverberation in the region without affecting relatively the local long-decay condition which produces the audible colouration.

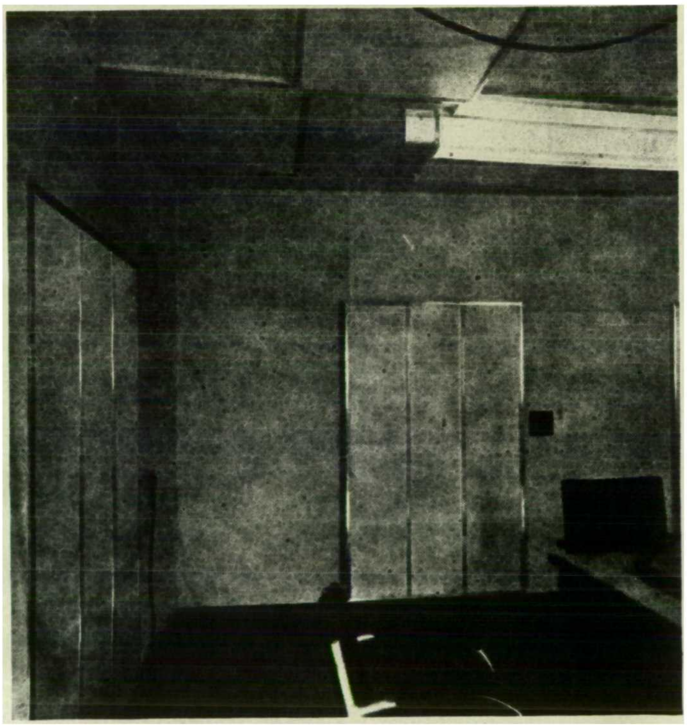

Fig. 7. Membrane absorbers with glass-fibre covers in a talks studio (3G Egton House) wall units tuned to 120 c/s (left hand wall) and 200 c/s (far wall). Recessed ceiling units tuned to 60 c/s and 90 c/s

The building construction of the absorbers is a fairly simple matter. They have usually been made in the form of projecting units of small frontal area and similar to that shown in Fig. 1, distributed around the walls and ceiling. The ad-

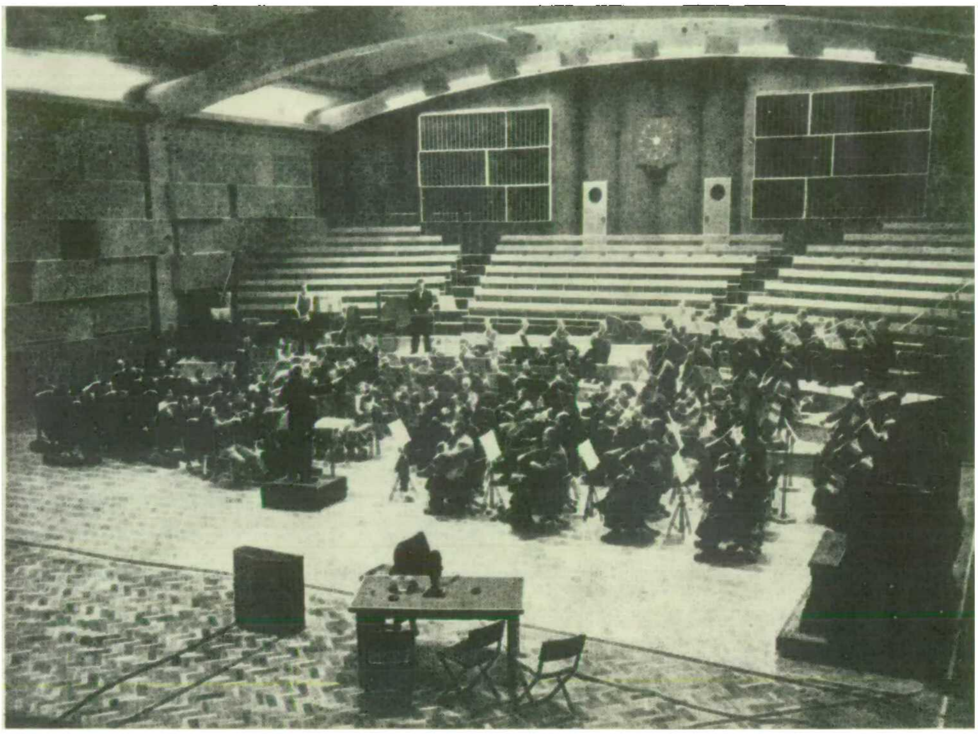

Fig. 8. View of Maida Vale Studio 1 showing membrane absorbers on left-hand wall. Perforated hardboard covers are fitted

vantages of this arrangement are that efficient absorption is obtained and that the rectangular projections thus formed on the wall surfaces improve the diffusion of sound in the studio at all frequencies. If additional diffusion is not required they may be recessed into the ceiling between the joists, or triangular-section units fitted along the angles between walls and ceiling make an extremely neat installation.

Whatever the operating frequency, it is usual to back the membrane with rockwool. This has the effect of broadening the bandwidth without seriously reducing the peak absorption, and thereby reduces the area required for a general reduction of reverberation time; for the higher frequencies a rockwool backing is, as already shown, essential.

Various covering materials have been used to conceal and protect the membrane. Materials such as glass-silk fabrics, hessian, etc. are used if additional high-frequency absorption is required. Fig. 7 shows a view of the Home News studio in Egton House, in which the absorbers are covered with glass-silk supported on expanded metal. The units seen on the walls are tuned to 120 c/s or 200 c/s while the ceiling units are recessed low-frequency absorbers (60 or 90 c/s). The most usual covering is of hardboard perforated in such a manner that the correct final absorption-frequency characteristic is obtained. The action of such perforated coverings has been described by Bolt.* Three types of perforation are in general use by the B.B.C.: (1) Hardboard perforated with holes amounting to ½ per cent. of the total area, which reflects all sound higher than approximately 300 c/s;

* Bolt, R. H. *Journal of Acoustical Society of America*, Vol 19, p. 917 (1947)

(2) 5 per cent. perforation reflecting above 700–1,000 c s; (3 Slots to 20 per cent. of the whole area which are transparent up to frequencies as high as 4,000 c s. Examples of absorbers with perforated coverings are shown in Fig 8, which is a photograph of the Symphony Orchestra Studio at Maida Vale. The absorbers, of three different depths and covered with perforated hardboard, will be seen along the side wall beyond the orchestra.

During the past four years membrane absorbers have been used in the acoustic treatment of sixteen studios. In every case they have been completely satisfactory and the absorption actually obtained has agreed closely with the predicted values. Provided that a rockwool backing is used their performance is very little affected by their frontal area or by the frequency at which maximum absorption occurs, and accurate design is consequently possible without recourse to complicated calculation or design charts.

The fact that the percentage bandwidth is independent of frequency, i.e., that the families of absorption curves are of almost the same shape on a logarithmic frequency scale, enables a single curve on transparent paper to be used for absorbers of all frequencies, the absorption peak being slid parallel to the frequency axis to the resonance point of the absorber chosen.